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FORM PTO-1390 U.S. DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE		ATTORNEY'S DOCKET NUMBER
TRANSMITTAL LETTER TO THE UNITED STATES DESIGNATED/ELECTED OFFICE (DO/EO/US) CONCERNING A FILING UNDER 35 U.S.C. 371		1454.1056/RAG 09/830413
INTERNATIONAL APPLICATION NO. PCT/DE99/03304	INTERNATIONAL FILING DATE 14 October 1999	PRIORITY DATE CLAIMED 27 October 1998

TITLE OF INVENTION
METHOD AND ARRANGEMENT FOR DETERMINING PARAMETERS OF A TECHNICAL SYSTEM

APPLICANT(S) FOR DO/EO/US
Dragan OBRADOVIC

Applicant herewith submits to the United States Designated/Elected Office (DO/EO/US) the following items and other information:

1. [X] This is a FIRST submission of items concerning a filing under 35 U.S.C. 371.
2. [X] This is an express request to immediately begin national examination procedures (35 U.S.C. 371(f)).
3. [ ] The US has been elected by the expiration of 19 months from the priority date (PCT Article 31).
4. [X] A copy of the International Application as filed (35 U.S.C. 371(c)(2))
    a. [X] is transmitted herewith (required only if not transmitted by the International Bureau).
    b. [ ] has been transmitted by the International Bureau.
    c. [ ] is not required, as the application was filed in the United States Receiving Office (RO/US).
5. [X] A translation of the International Application into English (35 U.S.C. 371(c)(2)).
6. [X] Amendments to the claims of the International Application under PCT Article 19 (35 U.S.C. 371(c)(3))
    a. [X] are transmitted herewith (required only if not transmitted by the International Bureau).
    b. [ ] have been transmitted by the International Bureau.
    c. [ ] is not required, as the application was filed in the United States Receiving Office (RO/US)
7. [X] A translation of the amendments to the claims under PCT Article 19 (35 U.S.C. 371(c)(3)).
8. [X] An oath or declaration of the inventor (35 U.S.C. 371(c)(4)).
9. [ ] A translation of the Annexes to the International Preliminary Examination Report under PCT Article 36 (35 U.S.C. 371(c)(5)).

Items 10-15 below concern document(s) or information included:

10. [X] An Information Disclosure Statement Under 37 CFR 1.97 and 1.98.
11. [X] An assignment document for recording.
    Please mail the recorded assignment document to:
    a. [X] the person whose signature, name & address appears at the bottom of this document.
    b. [ ] the following:
12. [X] A preliminary amendment.
13. [ ] A substitute specification
14. [ ] A change of power of attorney and/or address letter.
15. [X] Other items or information:
    International Preliminary Examination Report with International Search Report

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2. [X] The U.S. National Fee (35 U.S.C. 371(c)(1)) and other fees as follows:

CLAIMS	(1) FOR	(2) NUMBER FILED	(3) NUMBER EXTRA	(4) RATE	(5) CALCULATIONS
	TOTAL CLAIMS	16 -20=	0	x $ 18.00	0.00
	INDEPENDENT CLAIMS	2 -3=	0	x $ 80.00	0.00
	MULTIPLE DEPENDENT CLAIM(S) (if applicable)			+$270.00	0.00
	BASIC NATIONAL FEE (37 CFR 1.492(a)(1)-(4): [ ] Neither international preliminary examination fee (37 CFR 1.482) nor international search fee (37 CFR 1.445(a)(2)) paid to USPTO..............$1,000 [X] International preliminary examination fee (37 C.F.R. 1.482) not paid to USPTO but International Search Report prepared by the EPO or JPO.. . ...$ 860 [ ] International preliminary examination fee (37 C.F.R. 1.482) not paid to USPTO but international search fee (37 C.F.R. 1.445(a)(2) paid to USPTO..................$ 710 [ ] International preliminary examination fee paid to USPTO (37 CFR 1.482) but all claims did not satisfy provision of PCT Article 33(1)-(4).............$ 690 [ ] International preliminary examination fee paid to USPTO (37 CFR 1.482) and all claims satisfied provisions of PCT Article 33(2) to (4)................$ 100				860.00
	Surcharge of $130 for furnishing the National fee or oath or declaration later than [ ] 20 [ ] 30 mos. from the earliest claimed priority date (37 CFR 1.482(e)).				0.00
				TOTAL OF ABOVE CALCULATIONS	900.00
	Reduction by 1/2 for filing by small entity, if applicable. Affidavit must be filed also. (Note 37 CFR 1.9, 1.27, 1.28.)				
				SUBTOTAL	900.00
	Processing fee of $130 for furnishing the English Translation later than [ ] 20 [ ] 30 mos. from the earliest claimed priority date (37 CFR 1.482(f)).				.00
				TOTAL NATIONAL FEE	0.00
	Fee for recording the enclosed assignment (37 CFR 1.21(h)).				40.00
				TOTAL FEES ENCLOSED	940.00

a. [X] A check in the amount of $ 940.00 to cover the above fees is enclosed.

b. [ ] Please charge my Deposit Account No. 19-3935 in the Amount of $ to cover the above fees. A duplicate copy of this sheet is enclosed.

c. [X] The Commissioner is hereby authorized to charge any additional fees which may be required, or credit any overpayment to Deposit Account No. 19-3935. A duplicate copy of this sheet is enclosed.

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PATENT TRADEMARK OFFICE

4/27/01

DATE

Richard A. Gollhofer

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REGISTRATION NO. 31,106

GR 98 P 2958 1/PRTS

Description

Method and arrangement for determining parameters of a technical system

The invention relates to a method and an arrangement for determining parameters of a technical system.

During multichannel transmission and multichannel reception of signals, interference frequently occurs, for example, between the signals/images. One typical example in this case is mixing of a voice signal with noise, which can present a major problem in telecommunications and in video conferences. The present invention thus relates to the field of signal separation in order, for example, to recover an original voice signal.

Typical known techniques for separation of source signals based on mixed signals are based on time averaging or filtering of the signals. This intrinsically has disadvantages in terms of the computation complexity.

Methods based on so-called blind channel equalization (signal equalization without prior knowledge of the transmission channel) are also known, but these methods always require a certain amount of knowledge about the source signals, such as knowledge about their statistical distribution.

The problem of signal separation also occurs, for example, when two speakers are speaking into two microphones positioned at a distance from one another, so that each microphone receives a mixture of the signals spoken by the two speakers. The problem thus arises of separating the signal mixture once again, that is to say of separating a set of superimposed input signals. L. Molgedey, H.G. Schuster, "Separation of a Mixture of Independent Signals using Time-Delayed Correlations", Phys. Ref. Lett. 72, 3634 (1994) in this case discloses the following method: the problem of

separating n superimposed and correlated source signals (input signals) and at the same time of establishing mixing coefficients of the source intensities can be reduced to an intrinsic value problem, in which two symmetrical n x n matrices must be diagonalized simultaneously. The matrix elements are measurable time-delayed correlation functions. This intrinsic value problem can be solved by means of a neural network, in which case the learning rules for the lateral inhibiting interactions between the neurons can be established by means of a Liapunov function whose minima provide the (degenerate) solutions to the problem.

This method has also already been applied to the acoustic input signals (see F. Ehlers, H.G. Schuster, "Blind Separation of convolutive mixtures and an application in automatic speech recognition", IEEE Trans. Signal Proc. (1997).

DE 195 31 388 C1 discloses a signal separation method and a signal separation device for nonlinear mixtures of unknown signals (blind channel), which is illustrated schematically in Figure 3.

This German Patent relates to the separation of a signal mixture comprising the nonlinear superimposition of M unknown source signals X1, X2, where N ($N \geq M$) different mixtures of M source signals X1, X2 including any interference signal which may be present are supplied to a signal evaluation device, which analyzes the signal statistically to establish the nonlinear transmission factors and using these calculated factors to reverse the mixing process, so that the N outputs of the signal separation device contain, as approximately as possible, the M source signals without superimpositions.

It thus becomes possible to treat nonlinear mixtures, in which the case the term nonlinear means that the source signals X1, X2 are mixed by means of an unknown nonlinear system G. The unknown system G is described by a so-called Volterra series, and the signal separation device G-1 establishes the coefficients in the Volterra series. Once this

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SUBSTITUTE SPECIFICATION

TITLE OF THE INVENTION

System for determining parameters of a technical system

BACKGROUND OF THE INVENTION

FIELD OF THE INVENTION

The invention relates to a method and system for determining parameters of a technical system.

DESCRIPTION OF THE RELATED ART

During multichannel transmission and multichannel reception of signals, interference frequently occurs, for example, between the signals/images. One typical example in this case is mixing of a voice signal with noise, which can present a major problem in telecommunications and in video conferences. The present invention thus relates to the field of signal separation in order, for example, to recover an original voice signal.

Typical known techniques for separation of source signals based on mixed signals are based on time averaging or filtering of the signals. This intrinsically has disadvantages in terms of the computation complexity.

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coefficients of the source intensities can be reduced to an intrinsic value problem, in which two symmetrical n x n matrices must be diagonalized simultaneously. The matrix elements are measurable time-delayed correlation functions. This intrinsic value problem can be solved by a neural network, in which case the learning rules for the lateral inhibiting interactions between the neurons can be established by a Liapunov function whose minima provide the (degenerate) solutions to the problem.

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It thus becomes possible to treat nonlinear mixtures, in which case the term nonlinear means that the source signals X1, X2 are mixed by an unknown nonlinear system G. The unknown system G is described by a so-called Volterra series, and the signal separation device G-1 establishes the coefficients in the Volterra series. Once this is known, it is possible to unmix the signal mixture. Furthermore, the coefficients can be used for further analysis in order to determine the position or speed of the signal sources.

The method which is known from this document in this case essentially comprises two steps:

Firstly, the nonlinear equations which are selected uniquely by the selectable degree of nonlinearity in the mixing process are solved by a sliding time window, and the solutions are

averaged over this time. This time averaging process represents a major disadvantage of this known technique, since it increases the computation complexity, while at the same time increasing the time required for the calculation process.

Secondly, the potential formed from a sufficiently large number of different cumulants of the estimated output signals is minimized, with the values required to calculate the potential originating from a sliding time window whose length can be varied. In this case, it is assumed that the mixing system varies sufficiently slowly that this change can be ignored in the calculation of the sought mixing factors. According to this German Patent, the second said step is carried out by constructing and minimizing a cost function. When the global minimum is reached, the optimum values, in this case the transmission factors, have been found.

With regard to the time involved and the computation complexity, the method described in DE 195 31 388 C1 is disadvantageous, since the time averaging process has to be carried out at the end of the first method step mentioned above.

SUMMARY OF THE INVENTION

The present invention is thus based on the object of providing a method and system which allow the separation of superimposed, statistically mutually independent, acoustic signals with reduced computation complexity.

This object is achieved by a method for determining parameters of a technical system by determining output signals from a set of superimposed, statistically mutually independent input signals. The parameters are determined in such a manner that the statistical independence of the output signals is maximized.

A system for determining parameters of a technical system, in which output signals can be determined from a set of superimposed, statistically mutually independent input signals, has a processor that determines the parameters in such a manner that the statistical independence of the output signals is maximized. The parameters are preferably determined using an iterative method.

In a further refinement, the parameters are elements in an unmixing matrix, by which the set of superimposed input signals is multiplied or else convoluted, by which the output

signals are formed. The optimization of the parameters in the unmixing matrix is preferably obtained by the following steps:

- repetition of a time-delayed decorrelation calculation in order to determine the intrinsic values in the unmixing matrix,
- determination of the intrinsic values in the unmixing matrix for which cross-correlations assume a minimum value, and
- carrying out cumulant minimization, with the intrinsic values determined in the previous step being used as start values for the cumulant minimization.

The cumulant minimization can be used, for example, by training a neural network, or else by any other known minimization technique, such as gradient descent or Monte Carlo simulations.

In one development, at least one diagonal parameter of the unmixing matrix is set to a predetermined value during the optimization of the parameters in the unmixing matrix, thus ensuring the stability of the minimization process with respect to a global minimum.

The unmixing matrix is preferably limited to a finite impulse response, that is to say an FIR filter (Finite Impulse Response) is used to form the individual components of the unmixing matrix. The FIR filter may be either a causal FIR filter or else a non-causal FIR filter.

Furthermore, the unmixing matrix is preferably stabilized by projection on to a unit circle during the cumulant minimization process.

The developments apply not only to the method but also to a system in which a processor is set up in such a manner that the method can be carried out.

The invention and its developments can advantageously be used for separation of superimposed, statistically mutually independent input signals, in particular acoustic input signals.

The method and the system can be used for any desired number of input signals.

BRIEF DESCRIPTION OF THE DRAWINGS

These and other objects and advantages of the present invention will become more apparent and more readily appreciated from the following description of the preferred embodiments, taken in conjunction with the accompanying drawings, in which:

Figure 1 shows the use of a system for separation of superimposed, statistically mutually independent acoustic signals according to the exemplary embodiment,

Figure 2 shows a schematic illustration of the system from Figure 1, and

Figure 3 shows a signal separation device, which is known from the prior art (DE 195 31 388 C1), for nonlinear mixtures of unknown signals.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

Reference will now be made in detail to the preferred embodiments of the present invention, examples of which are illustrated in the accompanying drawings, wherein like reference numerals refer to like elements throughout.

The statistical independence between the source signals (the original voice signal and the noise), also referred to as input signals in the following text, is used to recover the original voice signal from a mixture of signals, and the inverse process to that of the dynamic system, which has resulted in the mixing of the signals, is trained essentially approximately (is learnt). Two different mixtures of the voice signal and of the noise signal are obtained, for example, by two microphones 1, 2 (see Figure 1) which are at a distance from one another and/or are aligned in opposite directions. The so-called time-delayed decorrelation technique (TDD) is used to initiate the learning phase in the method, that is to say in order to determine and specify start values for the learning phase, which allows the computation complexity for cumulant minimization as described in the following text to be reduced, and allows the risk of local minima to be reduced.

Figure 1 shows two microphones 1, 2, which pick up a first input signal Z1(t) and a second input signal Z2(t). These input signals Z1(t) and Z2(t) can in turn each be mixed with one another and with noise, as is represented symbolically by a mixing matrix S (see reference symbol 3) in Figure 1. After reception and/or transmission, a set X1(t) and X2(t) of

superimposed, statistically mutually independent input signals Z1(t) and Z2(t) is obtained. These signals are entered in a calculation unit 4, in which essentially two steps are carried out, as is represented symbolically by a calculation unit B (reference symbol 6) for the first step and a neural network 5 for the second step.

The calculation unit 4 determines two output signals Y1(t) and Y2(t), respectively, which are approximately equal to the input signals Z1(t) and Z2(t), respectively, when the parameters are set optimally in the calculation unit 4. In other words, when the parameters of the matrices which are used are set optimally in the calculation unit 4, this calculation unit 4 essentially carries out the inverse process to that of the dynamic mixing process, which is represented symbolically by the matrix S (reference symbol 3). The exemplary embodiment relates to the optimization process for setting the parameters for the unmixing matrix.

The parameters of the matrices in the calculation unit 4 are in this case optimized such that the statistical independence between the output signals Y1(t), Y2(t) obtained by the matrix process in the calculation unit 4 is maximized. To this end, the output signals Y1(t) and Y2(t), respectively, are fed back to the calculation unit 4 (see the feedback loops 7 and 8, respectively). An iterative method is used to determine whether the statistical independence of the output signal Y1(t) and Y2(t), respectively, has increased in comparison to the previous iteration step (so that the iteration is in the "right" direction, in the direction of the global minimum of a cost function, which will be described in the following text).

Figure 2 shows a mathematical representation of the layout from Figure 1, in which case the mixing process 3 can be described mathematically by a matrix S(q), and the unmixing process, which is intended to be carried out by the calculation unit 4, is symbolized by an unmixing matrix M(q).

Figure 2 thus illustrates the problem of separation of a so-called multichannel blind source (multiple channel source without a-priori knowledge) into two dimensions. In this case, it is assumed that the mixing system S(q), where q represents a unit delay, is stable and, at the same time, also has a stable inversion, that is to say it is a minimal phase system. Furthermore, it is assumed that the input signals Z1(t) and Z2(t) (for example a voice signal and a noise signal) are statistically mutually independent and do not have a Gaussian

distribution. The sets X1(t) and X2(t) of superimposed input signals Z1(t) and Z2(t) are input signals to an unmixing system having an unmixing matrix M(q) whose parameters (matrix elements) are trained to maximize the statistical independence between the output signals Y1(t) and Y2(t). In this case, the term "training" means the well known learning process of, for example, a neural network, which should be cited as an example of a technique to maximize the statistical independence. This is done by minimizing a cost function J(M), which will be described further below.

A cumulant cost function is formed, which minimizes the diagonal cumulant elements of the cumulant order 2-4:

$$\mathrm{Dcum} \approx J(M) = \sum_{i=1}^{4} \sum_{\mathrm{nondiag}} \left[c^{(i)}_{\mathrm{nondiag}}\right]^2$$

The following aspects of dynamic mixing by the mixing matrix S(q) need to be taken into account in this case:

- Stability of the unmixing system:

This is achieved by limiting M(q) to a finite impulse response (FIR filter). The stability of the FIR system M(q) can, furthermore, also be obtained by carrying out a projection on to a unit circle during the learning phase. Any non-causality of the inversion of S(q) which may be present can be compensated for by a suitable time shift (delay) to the input signal X(t).

- Uniqueness of the separated signals Y(t):

In the case of steady-state mixing processes, the original source signals are recovered by scaling. For dynamic unmixing, the risk of the separated signals Y(t) not being unique is even greater. It is obvious that, in the situation where Y1(t) and Y2(t) are statistically mutually independent, any linear-filtered modification of these signals will also still be statistically independent. Additional information is therefore required in order to reduce the inherent ambiguity of the problem.

- Gaussian deformation of the data:

Algorithms on a cumulant basis for steady-state blind source separation effectively minimize or eliminate higher-order diagonal cumulants corresponding to the output signals Y(t). On the other hand, linear filtering leads to the data being deformed to a Gaussian distribution, with the higher-order cumulants moving in the 0 direction. This can thus lead to limit solutions, in which the cost function reaches local minimum, but with the desired actual separation (global minimum) not being achieved. In order to avoid this undesirable situation, the structure of the unmixing transfer function (unmixing matrix) M(q) is subject to a number of limitations.

In order to avoid the abovementioned problems, an approach is chosen in which at least one (or else all) of the diagonal elements is or are set to the unit value:

M11(q) = 1 and M22(q) = 1.

This assumption is exact if the mixing elements S11(q) and/or S22(q) likewise have a unit value "1". Otherwise, it is assumed that S11(q) and/or S22(q) have stable inversions, which allows the diagonal elements to be scaled from M(q) to the unit value. This approach considerably reduces the ambiguity of a solution and, furthermore, effectively avoids the risk of excessive Gaussian-distribution deformation of the output signals. Even if, as stated above, the limitation to the diagonal elements of M(q) is at first glance to be highly restrictive, this assumption is generally satisfied in practical use. One typical example is the removal of noise from voice signals on the basis of a recording using two microphones, with the microphones being physically separated from one another or one microphone pointing in the direction of the speaker, while the other microphone points in the opposite direction, so that the second signal, facing away from the speaker, essentially includes only a noise signal.

The cumulant approach is based on the direct determination of the diagonal cumulant, as is stated in the article mentioned initially by F. Ehlers and H. Schuster, "Blind Separation of convolutive mixtures and an application in automatic speech recognition", IEEE Trans. Signal Proc. (1997), although this inherently has the disadvantage that numerical solution is highly complex. Suitable initialization is thus used for this minimization method. In order to determine start values for the minimization method, the present invention uses the technique of

time-delay decorrelation (TDD) for simultaneous decorrelation of two different time delays, in which case this TDD technique can be based on a suitable matrix intrinsic-value problem. As already stated, this TDD technique is used according to the present invention for initiation of the diagonal (cross-correlation) cumulant minimization problem.

In summary, the method can be subdivided into the two following steps:

1. Repetition of the TDD method on the basis of the intrinsic-value problem in the frequency domain for different delay pairs and determination of that solution for which the cross-correlation terms have a minimum value.

2. Initiation (start) of the diagonal cumulant minimization process on the basis of the start values (FIR parameters) determined in the above step.

A number of major characteristics and advantages will be summarized once again in the following text:

- No a-priori knowledge of the signal characteristics is required, with the exception of the necessity for statistical independence.

- The stability of the dynamic unmixing system is ensured by the modulation of its components as an FIR filter.

- Excessive Gaussian distribution deformation is avoided by the approach of at least one of the elements in the mixing transfer function matrix (unmixing matrix) being set to the unit value, or being able to be scaled to the unit value, and

- since the cumulant minimization step (step 2) requires a large amount of computation complexity, the learning algorithm, for example of a neural network, is initialized using the TDD method.

The following text contains a program in Matlab, e.g. Version 4, by which the exemplary embodiment described above can be implemented on a computer:

```
function { cost,out1,out2 ] } =cumulant_costFIRa2(par,input,p1,p11,p2,p22,a3,a4);
% { cost,out1,out2 ] } =cumulant_costFIRa2(par,input,p1,p11,p2,p22,a3,a4);
% cumulant cost
```

```
% FIR representation used
% filter function used in both directions (non_causal)

{ np,mp ] } =size(par);
fir1=par(1:p1);
fir11=par(1+p1:p1+p11);
fir2=par(1+p1+p11:p1+p11+p2);
fir22=par(p1+p11+p2+1:mp);
den=1;                                        %FIR only

out1= { input(:;1)-filter(fir1,den,input(:,2))-flipud(filter( { 0 fir11 ] } , { den ] }
,flipud(input(:,2)))) ] } ;
                          %/std(input(1,:));             %dlsim
%filter
out2= { input(:,2)-filter(fir2,den,input(:,1))-flipud(filter( { 0 fir22 ] } , ] }  { den],
flipud(input(:,1)))) ] } ;%/std(input(:,2));
                          %dlsim    %filter
out { out1 out2 ] } ;
%out1=out1/std(out1); % this scaling was not needed in examples in SIP98 paper
%out2=out2/std(out2);

Ld=0; % number of delays in calculating the cross-correlation
cost3=0;
cost4=0;
costALL1= {  ] } ;
costALL2= {  ] } ;
o12=out1.*out2;
cost2=mean(o12)^2;
```

```
o112=out1.*out1.*out2;
o122=out1.*out2.*out2;

if a3 ==1
        cost3= { mean(o122) ] } ^2+ { mean(o122) ] } ^2;
end

if a4 ==1

        cost4= { mean(o112.*out1)-3*mean(out1.^2)*mean(o12) ] } ^2+...
{ mean((out1.^2).*(out2.^2))-2*mean(o12)^2-
mean(out1.^2)*mean(out2.^2) ] } ^+...
{ mean(o122.*out2)-3*mean(out2.^2)*mean(o12) ] } ^2;
end

%cum4a= { cum4x(out1,out1,out1,out1) ] } ^2;
%cum4b= { cum4x(out2,out2,out2,out2) ] } ^2;
cost=cost2+a3*cost3+a4*cost4;                    %-cum4a-cum4b;
```

SUBSTITUTE ABSTRACT

SYSTEM FOR DETERMINING PARAMETERS OF A TECHNICAL SYSTEM

Parameters are established for a technical system, by means of which output signals can be determined from a set of superimposed, statistically mutually independent input signals. The parameters are determined in such a manner that the statistical independence of the output signals is maximized.

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Attorney Docket No. 1454.1056/RAG

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re Patent Application of:	)	
	)	
Dragan OBRADOVIC	)	
	)	Group Art Unit:
Application No.:	)	
	)	Examiner:
Filed: (concurrently)	)	
	)	

For: SYSTEM FOR DETERMINING PARAMETERS OF A TECHNICAL SYSTEM (as amended)

PRELIMINARY AMENDMENT

Assistant Commissioner for Patents
Washington, D.C. 20231

Sir:

Before examination of the above-identified application, please amend the application as follows:

IN THE TITLE

Change "METHOD AND ARRANGEMENT" to --SYSTEM--.

IN THE SPECIFICATION

Please REPLACE the pending specification with the SUBSTITUTE SPECIFICATION attached hereto.

IN THE ABSTRACT

Please REPLACE the originally filed Abstract with the enclosed Substitute Abstract.

IN THE CLAIMS

Please **CANCEL** claims 1-20 without prejudice or disclaimer of any of the subject matter claimed therein and **ADD** new claims in accordance with the following:

21. A method for determining parameters of a technical system to determine output signals from a set of superimposed, statistically mutually independent input signals, in which the parameters, which are elements in an unmixing matrix, by which the set of superimposed input signals are multiplied, and by which the output signals are formed, are determined by optimization of a statistical independence of the output signals, said method comprising:

repeatedly performing a time-delayed decorrelation calculation to determine intrinsic values in the unmixing matrix until cross-correlations are substantially minimized; and

carrying out cumulant minimization, with the intrinsic values determined by a final time-delayed decorrelation calculation being used as start values for the cumulant minimization.

22. The method as claimed in claim 21, in which the parameters are determined using an iterative method.

23. The method as claimed in claim 21, in which the cumulant minimization is carried out by training a neural network.

24. The method as claimed in claim 21, in which, during the optimization of the parameters of the unmixing matrix, at least one diagonal parameter in the unmixing matrix is set to a predetermined value.

25. The method as claimed in claim 21, in which the unmixing matrix is limited to a finite impulse response.

26. The method as claimed in claim 21, in which the unmixing matrix is stabilized by projection on to a unit circle during the cumulant minimization process.

27. The method as claimed in claim 21, used for separation of superimposed, statistically mutually independent input signals.

28. The method as claimed in claim 21, used for separation of superimposed, statistically mutually independent, acoustic input signals.

29. A system for determining parameters of a technical system to determine output signals from a set of superimposed, statistically mutually independent input signals, comprising:

a processor to determine the parameters, which are elements in an unmixing matrix, by which the set of superimposed input signals are multiplied, and by which the output signals are formed, by optimization of statistical independence of the output signals, through repetition of a time-delayed decorrelation calculation to determine intrinsic values in the unmixing matrix until cross-correlations are substantially minimized, and cumulant minimization, with the intrinsic values used as start values for the cumulant minimization.

30. The system as claimed in claim 29, in which the processor is set up in such a manner that the parameters are determined using an iterative method.

31. The system as claimed in claim 29, further comprising a neural network to perform the cumulant minimization after training.

32. The system as claimed in claim 9, in which the processor is set up in such a manner that, during the optimization of the parameters in the unmixing matrix, at least one diagonal parameter in the unmixing matrix is set to a predetermined value.

33. The system as claimed in claim 29, in which the processor is set up in such a manner that the unmixing matrix is limited to a finite impulse response.

34. The system as claimed in claim 29, in which the processor is set up in such a manner that the unmixing matrix is stabilized by projection on to a unit circle during the cumulant minimization process.

35. The system as claimed in claim 29, used for separation of superimposed, statistically mutually independent input signals.

36. The system as claimed in claim 29, used for separation of superimposed, statistically mutually independent, acoustic input signals.

REMARKS

This Preliminary Amendment is submitted to improve the form of the specification as originally-filed. It is respectfully requested that this Preliminary Amendment be entered in the above-referenced application.

In accordance with the foregoing, claims 1-20 have been canceled and claims 21-36 have been added. Thus, claims 21-36 are pending and are under consideration.

A substitute specification is also being filed herewith. The substitute specification is accompanied by a marked-up copy of the original specification. No new matter has been added.

If there are any questions regarding these matters, such questions can be addressed by telephone to the undersigned. Otherwise, an early action on the merits is respectfully solicited.

If any further fees are required in connection with the filing of this Preliminary Amendment, please charge same to our Deposit Account No. 19-3935.

Respectfully submitted,

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MARKED UP SPECIFICATION

[Description] TITLE OF THE INVENTION

[Method and arrangement] System for determining parameters of a technical system

BACKGROUND OF THE INVENTION

FIELD OF THE INVENTION

The invention relates to a method and [an arrangement] system for determining parameters of a technical system.

DESCRIPTION OF THE RELATED ART

During multichannel transmission and multichannel reception of signals, interference frequently occurs, for example, between the signals/images. One typical example in this case is mixing of a voice signal with noise, which can present a major problem in telecommunications and in video conferences. The present invention thus relates to the field of signal separation in order, for example, to recover an original voice signal.

Typical known techniques for separation of source signals based on mixed signals are based on time averaging or filtering of the signals. This intrinsically has disadvantages in terms of the computation complexity.

Methods based on so-called blind channel equalization (signal equalization without prior knowledge of the transmission channel) are also known, but these methods always require a certain amount of knowledge about the source signals, such as knowledge about their statistical distribution.

The problem of signal separation also occurs, for example, when two speakers are speaking into two microphones positioned at a distance from one another, so that each microphone receives a mixture of the signals spoken by the two speakers. The problem thus arises of separating the signal mixture once again, that is to say of separating a set of superimposed input signals. L. Molgedey, H.G. Schuster, "Separation of a Mixture of Independent Signals using Time-Delayed Correlations", Phys. Ref. Lett. 72, 3634 (1994) in this case discloses the following method: the problem of separating n superimposed and correlated source signals (input signals) and at the same time of establishing mixing

coefficients of the source intensities can be reduced to an intrinsic value problem, in which two symmetrical n x n matrices must be diagonalized simultaneously. The matrix elements are measurable time-delayed correlation functions. This intrinsic value problem can be solved by [means of] a neural network, in which case the learning rules for the lateral inhibiting interactions between the neurons can be established by [means of] a Liapunov function whose minima provide the (degenerate) solutions to the problem.

This method has also already been applied to the acoustic input signals (see F. Ehlers, H.G. Schuster, "Blind Separation of convolutive mixtures and an application in automatic speech recognition", IEEE Trans. Signal Proc. (1997).

DE 195 31 388 C1 discloses a signal separation method and a signal separation device for nonlinear mixtures of unknown signals (blind channel), which is illustrated schematically in Figure 3.

This German Patent relates to the separation of a signal mixture comprising the nonlinear superimposition of M unknown source signals X1, X2, where N ($N \geq M$) different mixtures of M source signals X1, X2 including any interference signal which may be present are supplied to a signal evaluation device, which analyzes the signal statistically to establish the nonlinear transmission factors and using these calculated factors to reverse the mixing process, so that the N outputs of the signal separation device contain, as approximately as possible, the M source signals without superimpositions.

It thus becomes possible to treat nonlinear mixtures, in which [the] case the term nonlinear means that the source signals X1, X2 are mixed by [means of] an unknown nonlinear system G. The unknown system G is described by a so-called Volterra series, and the signal separation device G-1 establishes the coefficients in the Volterra series. Once this is known, it is possible to unmix the signal mixture. Furthermore, the coefficients can be used for further analysis in order to determine the position or speed of the signal sources.

The method which is known from this document in this case essentially comprises two steps:

Firstly, the nonlinear equations which are selected uniquely by the selectable degree of nonlinearity in the mixing process are solved by a sliding time window, and the solutions are

averaged over this time. This time averaging process represents a major disadvantage of this known technique, since it increases the computation complexity, while at the same time increasing the time required for the calculation process.

Secondly, the potential formed from a sufficiently large number of different cumulants of the estimated output signals is minimized, with the values required to calculate the potential originating from a sliding time window whose length can be varied. In this case, it is assumed that the mixing system varies sufficiently slowly that this change can be ignored in the calculation of the sought mixing factors. According to this German Patent, the second said step is carried out by constructing and minimizing a cost function. When the global minimum is reached, the optimum values, in this case the transmission factors, have been found.

With regard to the time involved and the computation complexity, the method described in DE 195 31 388 C1 is disadvantageous, since the time averaging process has to be carried out at the end of the first method step mentioned above.

SUMMARY OF THE INVENTION

The present invention is thus based on the object of providing a method and [an arrangement] system which allow the separation of superimposed, statistically mutually independent, acoustic signals with reduced computation complexity.

This object is achieved by [the method and by the arrangement having the features according to the independent claims. In] a method for determining parameters of a technical system[,] by [means of which method] determining output signals [can be determined] from a set of superimposed, statistically mutually independent input signals. The [, the] parameters are determined in such a manner that the statistical independence of the output signals is maximized.

[An arrangement] A system for determining parameters of a technical system, [by means of] in which [system] output signals can be determined from a set of superimposed, statistically mutually independent input signals, has a processor [which is set up in such a manner] that determines the parameters [can be determined] in such a manner that the statistical independence of the output signals is maximized. [Advantageous developments of

the invention result from the dependent claims.] The parameters are preferably determined using an iterative method.

In a further refinement, the parameters are elements in an unmixing matrix, by which the set of superimposed input signals is multiplied or else convoluted, by which [means] the output signals are formed. The optimization of the parameters in the unmixing matrix is preferably obtained by the following steps:

- repetition of a time-delayed decorrelation calculation in order to determine the intrinsic values in the unmixing matrix,
- determination of the intrinsic values in the unmixing matrix for which cross-correlations assume a minimum value, and
- carrying out cumulant minimization, with the intrinsic values determined in the previous step being used as start values for the cumulant minimization.

The cumulant minimization can be used, for example, by training a neural network, or else by any other known minimization technique, such as gradient descent or Monte Carlo simulations.

In one development, at least one diagonal parameter of the unmixing matrix is set to a predetermined value during the optimization of the parameters in the unmixing matrix, thus ensuring the stability of the minimization process with respect to a global minimum.

The unmixing matrix is preferably limited to a finite impulse response, that is to say an FIR filter (Finite Impulse Response) is used to form the individual components of the unmixing matrix. The FIR filter may be either a causal FIR filter or else a non-causal FIR filter.

Furthermore, the unmixing matrix is preferably stabilized by projection on to a unit circle during the cumulant minimization process.

The developments apply not only to the method but also to [the arrangement,] a system in which [case, in the case of the arrangement, the] a processor is [in each case] set up in such a manner that the [corresponding] method [step] can be [or is] carried out.

The invention and its developments can advantageously be used for separation of superimposed, statistically mutually independent input signals, in particular acoustic input signals.

The method and the [arrangement] system can be used for any desired number of input signals.

[Further advantages, features and characteristics of the present invention will now be explained in more detail using an exemplary embodiment and with reference to the attached figures of]

BRIEF DESCRIPTION OF THE DRAWINGS

These and other objects and advantages of the present invention will become more apparent and more readily appreciated from the following description of the preferred embodiments, taken in conjunction with the accompanying drawings, in which:

Figure 1 shows the use of a system for separation of superimposed, statistically mutually independent acoustic signals according to the exemplary embodiment,

Figure 2 shows a schematic illustration of the system from Figure 1, and

Figure 3 shows a signal separation device, which is known from the prior art (DE 195 31 388 C1), for nonlinear mixtures of unknown signals.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

Reference will now be made in detail to the preferred embodiments of the present invention, examples of which are illustrated in the accompanying drawings, wherein like reference numerals refer to like elements throughout.

The statistical independence between the source signals (the original voice signal and the noise), also referred to as input signals in the following text, is used to recover the original voice signal from a mixture of signals, and the inverse process to that of the dynamic system, which has resulted in the mixing of the signals, is trained essentially approximately (is learnt). Two different mixtures of the voice signal and of the noise signal are obtained, for example, by [means of] two microphones 1, 2 (see Figure 1) which are at a distance from one another and/or are aligned in opposite directions. The so-called time-delayed decorrelation technique (TDD) is used to initiate the learning phase in the method, that is to say in order to determine and specify start values for the learning phase, which allows the computation complexity for

cumulant minimization as described in the following text to be reduced, and allows the risk of local minima to be reduced.

Figure 1 shows two microphones 1, 2, which pick up a first input signal Z1(t) and a second input signal Z2(t). These input signals Z1(t) and Z2(t) can in turn each be mixed with one another and with noise, as is represented symbolically by a mixing matrix S (see reference symbol 3) in Figure 1. After reception and/or transmission, a set X1(t) and X2(t) of superimposed, statistically mutually independent input signals Z1(t) and Z2(t) is obtained. These signals are entered in a calculation unit 4, in which essentially two steps are carried out, as is represented symbolically by a calculation unit B (reference symbol 6) for the first step and a neural network 5 for the second step.

The calculation unit 4 determines two output signals Y1(t) and Y2(t), respectively, which are approximately equal to the input signals Z1(t) and Z2(t), respectively, when the parameters are set optimally in the calculation unit 4. In other words, when the parameters of the matrices which are used are set optimally in the calculation unit 4, this calculation unit 4 essentially carries out the inverse process to that of the dynamic mixing process, which is represented symbolically by the matrix S (reference symbol 3). The exemplary embodiment relates to the optimization process for setting the parameters for the unmixing matrix.

The parameters of the matrices in the calculation unit 4 are in this case optimized such that the statistical independence between the output signals Y1(t), Y2(t) obtained by the matrix process in the calculation unit 4 is maximized. To this end, the output signals Y1(t) and Y2(t), respectively, are fed back to the calculation unit 4 (see the feedback loops 7 and 8, respectively). An iterative method is used to determine whether the statistical independence of the output signal Y1(t) and Y2(t), respectively, has increased in comparison to the previous iteration step (so that the iteration is in the "right" direction, in the direction of the global minimum of a cost function, which will be described in the following text)[, or not].

Figure 2 shows a mathematical representation of the layout from Figure 1, in which case the mixing process 3 can be described mathematically by a matrix S(q), and the unmixing process, which is intended to be carried out by the calculation unit 4, is symbolized by an unmixing matrix M(q).

Figure 2 thus illustrates the problem of separation of a so-called multichannel blind source (multiple channel source without a-priori knowledge) into two dimensions. In this case, it is assumed that the mixing system S(q), where q represents a unit delay, is stable and, at the same time, also has a stable inversion, that is to say it is a minimal phase system. Furthermore, it is assumed that the input signals Z1(t) and Z2(t) (for example a voice signal and a noise signal) are statistically mutually independent and do not have a Gaussian distribution. The sets X1(t) and X2(t) of superimposed input signals Z1(t) and Z2(t) are input signals to an unmixing system having an unmixing matrix M(q) whose parameters (matrix elements) are trained to maximize the statistical independence between the output signals Y1(t) and Y2(t). In this case, the term "training" means the well known learning process of, for example, a [neuron] <u>neural</u> network, which should be cited as an example of a technique to maximize the statistical independence. This is done by minimizing a cost function J(M), which will be described further below.

A cumulant cost function is formed, which minimizes the diagonal cumulant elements of the cumulant order 2-4:

$$\mathrm{Dcum} \approx J(M) = \sum_{i=1}^{4} \sum_{\mathrm{nondiag}} \left[c^{(i)}\mathrm{nondiag}\right]^2$$

The following aspects of dynamic mixing by the mixing matrix S(q) need to be taken into account in this case:

- Stability of the unmixing system:

This is achieved by limiting M(q) to a finite impulse response (FIR filter). The stability of the FIR system M(q) can, furthermore, also be obtained by carrying out a projection on to a unit circle during the learning phase. Any non-causality of the inversion of S(q) which may be present can be compensated for by a suitable time shift (delay) to the input signal X(t).

- Uniqueness of the separated signals Y(t):

In the case of steady-state mixing processes, the original source signals are recovered by scaling. For dynamic unmixing, the risk of the separated signals Y(t) not being unique is even greater. It is obvious that, in the situation where Y1(t) and Y2(t) are statistically mutually independent, any linear-filtered modification of these signals will also still be statistically independent. Additional information is therefore required in order to reduce the inherent ambiguity of the problem.

- Gaussian deformation of the data:

Algorithms on a cumulant basis for steady-state blind source separation effectively minimize or eliminate higher-order diagonal cumulants corresponding to the output signals Y(t). On the other hand, linear filtering leads to the data being deformed to a Gaussian distribution, with the higher-order cumulants moving in the 0 direction. This can thus lead to limit solutions, in which the cost function reaches local minimum, but with the desired actual separation (global minimum) not being achieved. In order to avoid this undesirable situation, the structure of the unmixing transfer function (unmixing matrix) M(q) is subject to a number of limitations.

In order to avoid the abovementioned problems, an approach is chosen in which at least one (or else all) of the diagonal elements is or are set to the unit value:

M11(q) = 1 and M22(q) = 1.

This assumption is exact if the mixing elements S11(q) and/or S22(q) likewise have a unit value "1". Otherwise, it is assumed that S11(q) and/or S22(q) have stable inversions, which allows the diagonal elements to be scaled from M(q) to the unit value. This approach considerably reduces the ambiguity of a solution and, furthermore, effectively avoids the risk of excessive Gaussian-distribution deformation of the output signals. Even if, as stated above, the limitation to the diagonal elements of M(q) is at first glance to be highly restrictive, this assumption is generally satisfied in practical use. One typical example is the removal of noise from voice signals on the basis of a recording using two microphones, with the microphones being physically separated from one another or one microphone pointing in the direction of the

speaker, while the other microphone points in the opposite direction, so that the second signal, facing away from the speaker, essentially includes only a noise signal.

The cumulant approach is based on the direct determination of the diagonal cumulant, as is stated in the article mentioned initially by F. Ehlers and H. Schuster, "Blind Separation of convolutive mixtures and an application in automatic speech recognition", IEEE Trans. Signal Proc. (1997), although this inherently has the disadvantage that numerical solution is highly complex. Suitable initialization is thus used for this minimization method. In order to determine start values for the minimization method, the present invention uses the technique of time-delay decorrelation (TDD) for simultaneous decorrelation of two different time delays, in which case this TDD technique can be based on a suitable matrix intrinsic-value problem. As already stated, this TDD technique is used according to the present invention for initiation of the diagonal (cross-correlation) cumulant minimization problem.

In summary, the method can be subdivided into the two following steps:

1. Repetition of the TDD method on the basis of the intrinsic-value problem in the frequency domain for different delay pairs and determination of that solution for which the cross-correlation terms have a minimum value.

2. Initiation (start) of the diagonal cumulant minimization process on the basis of the start values (FIR parameters) determined in the above step.

A number of major characteristics and advantages will be summarized once again in the following text:

- No a-priori knowledge of the signal characteristics is required, with the exception of the necessity for statistical independence.

- The stability of the dynamic unmixing system is ensured by the modulation of its components as an FIR filter.

- Excessive Gaussian distribution deformation is avoided by the approach of at least one of the elements in the mixing transfer function matrix (unmixing matrix) being set to the unit value, or being able to be scaled to the unit value, and

- since the cumulant minimization step (step 2) requires a large amount of computation complexity, the learning algorithm, for example of a neural network, is initialized using the TDD method.

The following text contains a program in Matlab, <u>e.g.</u> Version 4 [or Version] [lacuna], by [means of] which the exemplary embodiment described above can be implemented on a computer:

```
function[ [ ] { cost,out1,out2 [ ] ] } =cumulant_costFIRa2(par,input,p1,p11,p2,p22,a3,a4);
%[ [ ] { cost,out1,out2 [ ] ] } =cumulant_costFIRa2(par,input,p1,p11,p2,p22,a3,a4);
% cumulant cost
% FIR representation used
% filter function used in both directions (non_causal)

[ [ ] { np,mp [ ] ] } =size(par);
fir1=par(1:p1);
fir11=par(1+p1:p1+p11);
fir2=par(1+p1+p11:p1+p11+p2);
fir22=par(p1+p11+p2+1:mp);
den=1;                                    %FIR only

out1=[ [ ] { input(:;1)-filter(fir1,den,input(:,2))-flipud(filter([ [ ] { 0 fir11 [ ] ] } , [ [ ] { den [
] ] } ,flipud(input(:,2)))) [ ] ] } ;
                          %/std(input(1,:));                    %dlsim
%filter
out2=[ [ ] { input(:,2)-filter(fir2,den,input(:,1))-flipud(filter([ [ ] { 0 fir22 [ ] ] } , [ [  [ ] ] } {
den],
flipud(input(:,1)))) [ ] ] } ;%/std(input(:,2));
                          %dlsim     %filter
out[ [ ] { out1 out2 [ ] ] } ;
```

```
%out1=out1/std(out1); % this scaling was not needed in examples in SIP98 paper
%out2=out2/std(out2);

Ld=0; % number of delays in calculating the cross-correlation
cost3=0;
cost4=0;
costALL1=[ [ ] { [ ] ] } ;
costALL2=[ [ ] { [ ] ] } ;
o12=out1.*out2;
cost2=mean(o12)^2;

o112=out1.*out1.*out2;
o122=out1.*out2.*out2;

if a3 ==1
        cost3=[ [ ] { mean(o122) [ ] ] } ^2+[ [ ] { mean(o122) [ ] ] } ^2;
end

if a4 ==1

        cost4=[ [ ] { mean(o112.*out1)-3*mean(out1.^2)*mean(o12) [ ] ] } ^2+...
[ [ ] { mean((out1.^2).*(out2.^2))-2*mean(o12)^2-
mean(out1.^2)*mean(out2.^2) [ ] ] } ^+...
[ [ ] { mean(o122.*out2)-3*mean(out2.^2)*mean(o12) [ ] ] } ^2;
end

%cum4a=[ [ ] { cum4x(out1,out1,out1,out1) [ ] ] } ^2;
%cum4b=[ [ ] { cum4x(out2,out2,out2,out2) [ ] ] } ^2;
cost=cost2+a3*cost3+a4*cost4;                    %-cum4a-cum4b;
```

is known, it is possible to unmix the signal mixture. Furthermore, the coefficients can be used for further analysis in order to determine the position or speed of the signal sources.

The method which is known from this document in this case essentially comprises two steps:

- Firstly, the nonlinear equations which are selected uniquely by the selectable degree of nonlinearity in the mixing process are solved by a sliding time window, and the solutions are averaged over this time. This time averaging process represents a major disadvantage of this known technique, since it increases the computation complexity, while at the same time increasing the time required for the calculation process.
- Secondly, the potential formed from a sufficiently large number of different cumulants of the estimated output signals is minimized, with the values required to calculate the potential originating from a sliding time window whose length can be varied. In this case, it is assumed that the mixing system varies sufficiently slowly that this change can be ignored in the calculation of the sought mixing factors. According to this German Patent, the second said step is carried out by constructing and minimizing a cost function. When the global minimum is reached, the optimum values, in this case the transmission factors, have been found.

With regard to the time involved and the computation complexity, the method described in DE 195 31 388 C1 is disadvantageous, since the time averaging process has to be carried out at the end of the first method step mentioned above.

The present invention is thus based on the object of providing a method and an arrangement which allow the separation of superimposed, statistically mutually independent, acoustic signals with reduced computation complexity.

This object is achieved by the method and by the arrangement having the features according to the independent claims.

In a method for determining parameters of a technical system, by means of which method output signals can be determined from a set of superimposed, statistically mutually independent input signals, the parameters are determined in such a manner that the statistical independence of the output signals is maximized.

An arrangement for determining parameters of a technical system, by means of which system output signals can be determined from a set of superimposed, statistically mutually independent input signals, has a processor which is set up in such a manner that the parameters can be determined in such a manner that the statistical independence of the output signals is maximized.

Advantageous developments of the invention result from the dependent claims.

The parameters are preferably determined using an iterative method.

In a further refinement, the parameters are elements in an unmixing matrix, by which the set of superimposed input signals is multiplied or else convoluted, by which means the output signals are formed.

The optimization of the parameters in the unmixing matrix is preferably obtained by the following steps:

- repetition of a time-delayed decorrelation calculation in order to determine the intrinsic values in the unmixing matrix,

- determination of the intrinsic values in the unmixing matrix for which cross-correlations assume a minimum value, and
- carrying out cumulant minimization, with the intrinsic values determined in the previous step being used as start values for the cumulant minimization.

The cumulant minimization can be used, for example, by training a neural network, or else by any other known minimization technique, such as gradient descent or Monte Carlo simulations.

In one development, at least one diagonal parameter of the unmixing matrix is set to a predetermined value during the optimization of the parameters in the unmixing matrix, thus ensuring the stability of the minimization process with respect to a global minimum.

The unmixing matrix is preferably limited to a finite impulse response, that is to say an FIR filter (Finite Impulse Response) is used to form the individual components of the unmixing matrix. The FIR filter may be either a causal FIR filter or else a non-causal FIR filter.

Furthermore, the unmixing matrix is preferably stabilized by projection on to a unit circle during the cumulant minimization process.

The developments apply not only to the method but also to the arrangement, in which case, in the case of the arrangement, the processor is in each case set up in such a manner that the corresponding method step can be or is carried out.

The invention and its developments can advantageously be used for separation of superimposed, statistically mutually independent input signals, in particular acoustic input signals.

The method and the arrangement can be used for any desired number of input signals.

Further advantages, features and characteristics of the present invention will now be explained in more detail using an exemplary embodiment and with reference to the attached figures of the drawings, in which:

Figure 1 shows the use of a system for separation of superimposed, statistically mutually independent acoustic signals according to the exemplary embodiment,

Figure 2 shows a schematic illustration of the system from Figure 1, and

Figure 3 shows a signal separation device, which is known from the prior art (DE 195 31 388 C1), for nonlinear mixtures of unknown signals.

The statistical independence between the source signals (the original voice signal and the noise), also referred to as input signals in the following text, is used to recover the original voice signal from a mixture of signals, and the inverse process to that of the dynamic system, which has resulted in the mixing of the signals, is trained essentially approximately (is learnt). Two different mixtures of the voice signal and of the noise signal are obtained, for example, by means of two microphones 1, 2 (see **Figure 1**) which are at a distance from one another and/or are aligned in opposite directions. The so-called time-delayed decorrelation technique (TDD) is used to initiate the learning phase in the method, that is to say in order to determine and specify start values for the learning phase, which allows the computation complexity for cumulant minimization as described in the following text to be reduced, and allows the risk of local minima to be reduced.

Figure 1 shows two microphones 1, 2, which pick up a first input signal Z1(t) and a second input signal Z2(t). These input signals Z1(t) and Z2(t) can in turn each be mixed with one another and with noise, as is represented symbolically by a mixing matrix S (see reference symbol 3) in **Figure 1**. After reception and/or transmission, a set X1(t) and X2(t) of superimposed, statistically mutually independent input signals Z1(t) and Z2(t) is obtained. These signals are entered in a calculation unit 4, in which essentially two steps are carried out, as is represented symbolically by a calculation unit B (reference symbol 6) for the first step and a neural network 5 for the second step.

The calculation unit 4 determines two output signals Y1(t) and Y2(t), respectively, which are approximately equal to the input signals Z1(t) and Z2(t), respectively, when the parameters are set optimally in the calculation unit 4. In other words, when the parameters of the matrices which are used are set optimally in the calculation unit 4, this calculation unit 4 essentially carries out the inverse process to that of the dynamic mixing process, which is represented symbolically by the matrix S (reference symbol 3). The exemplary embodiment relates to the optimization process for setting the parameters for the unmixing matrix.

The parameters of the matrices in the calculation unit 4 are in this case optimized such that the statistical independence between the output signals Y1(t), Y2(t) obtained by the matrix process in the calculation unit 4 is maximized. To this end, the output signals Y1(t) and Y2(t), respectively, are fed back to the calculation unit 4 (see the feedback loops 7 and 8, respectively). An iterative method is used to determine whether the statistical independence of the output signal Y1(t) and Y2(t),

respectively, has increased in comparison to the previous iteration step (so that the iteration is in the "right" direction, in the direction of the global minimum of a cost function, which will be described in the following text), or not.

Figure 2 shows a mathematical representation of the layout from **Figure 1**, in which case the mixing process 3 can be described mathematically by a matrix S(q), and the unmixing process, which is intended to be carried out by the calculation unit 4, is symbolized by an unmixing matrix M(q).

Figure 2 thus illustrates the problem of separation of a so-called multichannel blind source (multiple channel source without a-priori knowledge) into two dimensions. In this case, it is assumed that the mixing system S(q), where q represents a unit delay, is stable and, at the same time, also has a stable inversion, that is to say it is a minimal phase system. Furthermore, it is assumed that the input signals Z1(t) and Z2(t) (for example a voice signal and a noise signal) are statistically mutually independent and do not have a Gaussian distribution. The sets X1(t) and X2(t) of superimposed input signals Z1(t) and Z2(t) are input signals to an unmixing system having an unmixing matrix M(q) whose parameters (matrix elements) are trained to maximize the statistical independence between the output signals Y1(t) and Y2(t). In this case, the term "training" means the well known learning process of, for example, a neuron network, which should be cited as an example of a technique to maximize the statistical independence. This is done by minimizing a cost function J(M), which will be described further below.

A cumulant cost function is formed, which minimizes the diagonal cumulant elements of the cumulant order 2-4:

$$\mathrm{Dcum} \approx J(M) = \sum_{i=1}^{4} \sum_{\mathrm{nondiag}} \left[c^{(i)}\mathrm{nondiag}\right]^2$$

The following aspects of dynamic mixing by the mixing matrix S(q) need to be taken into account in this case:

- Stability of the unmixing system:
  This is achieved by limiting M(q) to a finite impulse response (FIR filter). The stability of the FIR system M(q) can, furthermore, also be obtained by carrying out a projection on to a unit circle during the learning phase. Any non-causality of the inversion of S(q) which may be present can be compensated for by a suitable time shift (delay) to the input signal X(t).

- Uniqueness of the separated signals Y(t):
  In the case of steady-state mixing processes, the original source signals are recovered by scaling. For dynamic unmixing, the risk of the separated signals Y(t) not being unique is even greater. It is obvious that, in the situation where Y1(t) and Y2(t) are statistically mutually independent, any linear-filtered modification of these signals will also still be statistically independent. Additional information is therefore required in order to reduce the inherent ambiguity of the problem.

- Gaussian deformation of the data:
  Algorithms on a cumulant basis for steady-state blind source separation effectively minimize or eliminate higher-order diagonal cumulants corresponding to the output signals Y(t). On the other hand, linear filtering leads to the data being deformed to a Gaussian distribution, with the higher-order cumulants moving in the 0 direction. This can thus lead to limit solutions, in which the cost function reaches local minimum, but with the desired actual separation (global minimum) not being achieved. In order to avoid this undesirable situation, the structure of the unmixing transfer function (unmixing matrix) M(q) is subject to a number of limitations.

In order to avoid the abovementioned problems, an approach is chosen in which at least one (or else all) of the diagonal elements is or are set to the unit value:

M11(q) = 1

and

M22(q) = 1.

This assumption is exact if the mixing elements S11(q) and/or S22(q) likewise have a unit value "1". Otherwise, it is assumed that S11(q) and/or S22(q) have stable inversions, which allows the diagonal elements to be scaled from M(q) to the unit value. This approach considerably reduces the ambiguity of a solution and, furthermore, effectively avoids the risk of excessive Gaussian-distribution deformation of the output signals. Even if, as stated above, the limitation to the diagonal elements of M(q) is at first glance to be highly restrictive, this assumption is generally satisfied in practical use. One typical example is the removal of noise from voice signals on the basis of a recording using two microphones, with the microphones being physically separated from one another or one microphone pointing in the direction of the speaker, while the other microphone points in the opposite direction, so that the second signal, facing away from the speaker, essentially includes only a noise signal.

The cumulant approach is based on the direct determination of the diagonal cumulant, as is stated in the article mentioned initially by F. Ehlers and H. Schuster, "Blind Separation of convolutive mixtures and an application in automatic speech recognition", IEEE Trans. Signal Proc. (1997), although this inherently has the disadvantage that numerical solution is highly complex. Suitable initialization is thus used for this minimization method. In order to determine start values

for the minimization method, the present invention uses the technique of time-delay decorrelation (TDD) for simultaneous decorrelation of two different time delays, in which case this TDD technique can be based on a suitable matrix intrinsic-value problem. As already stated, this TDD technique is used according to the present invention for initiation of the diagonal (cross-correlation) cumulant minimization problem.

In summary, the method can be subdivided into the two following steps:

1. Repetition of the TDD method on the basis of the intrinsic-value problem in the frequency domain for different delay pairs and determination of that solution for which the cross-correlation terms have a minimum value.

2. Initiation (start) of the diagonal cumulant minimization process on the basis of the start values (FIR parameters) determined in the above step.

A number of major characteristics and advantages will be summarized once again in the following text:

- No a-priori knowledge of the signal characteristics is required, with the exception of the necessity for statistical independence.
- The stability of the dynamic unmixing system is ensured by the modulation of its components as an FIR filter.
- Excessive Gaussian distribution deformation is avoided by the approach of at least one of the elements in the mixing transfer function matrix (unmixing matrix) being set to the unit value, or being able to be scaled to the unit value, and
- since the cumulant minimization step (step 2) requires a large amount of computation complexity,

the learning algorithm, for example of a neural network, is initialized using the TDD method.

The following text contains a program in Matlab, Version 4 or Version [lacuna], by means of which the exemplary embodiment described above can be implemented on a computer:

```
function[cost,out1,out2]=cumulant_costFIRa2(par,input,p1,p11,p2,p
22,a3,a4);
%[cost,out1,out2]=cumulant_costFIRa2(par,input,p1,p11,p2,p22,a3,a
4);
% cumulant cost
% FIR representation used
% filter function used in both directions (non_causal)

[np,mp]=size(par);
fir1=par(1:p1);
fir11=par(1+p1:p1+p11);
fir2=par(1+p1+p11:p1+p11+p2);
fir22=par(p1+p11+p2+1:mp);
den=1;                                    %FIR only

out1=[input(:;1)-filter(fir1,den,input(:,2))-flipud(filter([0
fir11], [den],flipud(input(:,2))))];
                        %/std(input(1,:));          %dlsim

%filter
out2=[input(:,2)-filter(fir2,den,input(:,1))-flipud(filter([0
fir22], [den],
flipud(input(:,1))))];%/std(input(:,2));
                        %dlsim    %filter

out[out1 out2];
%out1=out1/std(out1); % this scaling was not needed in examples
in SIP98 paper
%out2=out2/std(out2);

Ld=0; % number of delays in calculating the cross-correlation
cost3=0;
cost4=0;
costALL1=[];
```

```
costALL2=[];
o12=out1.*out2;
cost2=mean(o12)^2;

o112=out1.*out1.*out2;
o122=out1.*out2.*out2;

if a3 ==1
      cost3=[mean(o122)]^2+[mean(o122)]^2;

end

if a4 ==1

      cost4=[mean(o112.*out1)-
3*mean(out1.^2)*mean(o12)]^2+...
[mean((out1.^2).*(out2.^2))-2*mean(o12)^2-
mean(out1.^2)*mean(out2.^2)]^+...
[mean(o122.*out2)-3*mean(out2.^2)*mean(o12)]^2;

end

%cum4a=[cum4x(out1,out1,out1,out1)]^2;
%cum4b=[cum4x(out2,out2,out2,out2)]^2;

cost=cost2+a3*cost3+a4*cost4;              %-cum4a-cum4b;
```

Article 34

Patent Claims

1. A method for determining parameters of a technical system, by means of which output signals can be determined from a set of superimposed, statistically mutually independent input signals, in which the parameters, which are elements in an unmixing matrix, by which the set of superimposed input signals are multiplied, and by which means the output signals are formed, are determined by optimization of a statistical independence of the output signals, using the following steps:

- repetition of a time-delayed decorrelation calculation (6) in order to determine the intrinsic values in the unmixing matrix,
- determination of the intrinsic values in the unmixing matrix for which cross-correlations assume a minimum value, and
- carrying out cumulant minimization (5), with the intrinsic values determined in the previous step being used as start values for the cumulant minimization.

2. The method as claimed in claim 1, in which the parameters are determined using an iterative method.

3. The method as claimed in claim 1 or 2, in which the cumulant minimization is carried out by training a neural network (5).

4. The method as claimed in one of claims 1 to 3, in which, during the optimization of the parameters of the unmixing matrix, at least one diagonal parameter in the unmixing matrix is set to a predetermined value.

5. The method as claimed in one of claims 1 to 4, in which the unmixing matrix is limited to a finite impulse response.

6. The method as claimed in one of claims 1 to 5, in which the unmixing matrix is stabilized by projection on to a unit circle during the cumulant minimization process (5).

7. The method as claimed in one of claims 1 to 6, used for separation of superimposed, statistically mutually independent input signals.

8. The method as claimed in one of claims 1 to 6, used for separation of superimposed, statistically mutually independent, acoustic input signals.

9. An arrangement for determining parameters of a technical system, by means of which output signals can be determined from a set of superimposed, statistically mutually independent input signals, having a processor which is set in such a manner that the parameters, which are elements in an unmixing matrix, by which the set of superimposed input signals are multiplied, and by which means the output signals are formed, are determined by optimization of a statistical independence of the output signals, using the following steps:

- repetition of a time-delayed decorrelation calculation (6) in order to determine the intrinsic values in the unmixing matrix,
- determination of the intrinsic values in the unmixing matrix for which cross-correlations assume a minimum value, and
- carrying out cumulant minimization (5), with the intrinsic values determined in the previous step being used as start values for the cumulant minimization.

10. The arrangement as claimed in claim 9, in which the processor is set up in such a manner that the parameters are determined using an iterative method.

11. The arrangement as claimed in claim 9 or 10, in which the processor is set up in such a manner that the cumulant minimization is carried out by training a neural network (5).

12. The arrangement as claimed in one of claims 9 to 11,
in which the processor is set up in such a manner that, during the optimization of the parameters in the

unmixing matrix, at least one diagonal parameter in the unmixing matrix is set to a predetermined value.

13. The arrangement as claimed in one of claims 9 to 12,
in which the processor is set up in such a manner that the unmixing matrix is limited to a finite impulse response.

14. The arrangement as claimed in one of claims 9 to 13, in which the processor is set up in such a manner that the unmixing matrix is stabilized by projection on to a unit circle during the cumulant minimization process (5).

15. The arrangement as claimed in one of claims 9 to 14, used for separation of superimposed, statistically mutually independent input signals.

16. The arrangement as claimed in one of claims 9 to 14, used for separation of superimposed, statistically mutually independent, acoustic input signals.

Abstract

Method and arrangement for determining parameters of a technical system

Parameters are established for a technical system, by means of which output signals can be determined from a set of superimposed, statistically mutually independent input signals. The parameters are determined in such a manner that the statistical independence of the output signals is maximized.

Significant figure, Figure 1

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0 **0-1**	**Vom Anmeldeamt auszufüllen** Internationales Aktenzeichen.	
0-2	Internationales Anmeldedatum	
0-3	Name des Anmeldeamts und "PCT International Application"	

0-4 0-4-1	**Formular - PCT/RO/101 PCT-Antrag** erstellt durch Benutzung von	PCT-EASY Version 2.84 (aktualisiert 01.07.1999)
0-5	**Antragsersuchen** Der Unterzeichnete beantragt, daß die vorliegende internationale Anmeldung nach dem Vertrag über die internationale Zusammenarbeit auf dem Gebiet des Patentwesens behandelt wird	
0-6	**(Vom Anmelder gewähltes) Anmeldeamt**	Deutsches Patent- und Markenamt (RO/DE)
0-7	**Aktenzeichen des Anmelders oder Anwalts**	GR98P2958P
I	**Bezeichnung der Erfindung**	VERFAHREN UND ANORDNUNG ZUR ERMITTLUNG VON PARAMETERN EINES TECHNISCHEN SYSTEMS
II	**Anmelder**	
II-1	Diese Person ist	nur Anmelder
II-2	Anmelder für	Alle Bestimmungsstaaten mit Ausnahme von US
II-4	Name	SIEMENS AKTIENGESELLSCHAFT
II-5	Anschrift:	Wittelsbacherplatz 2 D-80333 München Deutschland
II-6	Staatsangehörigkeit (Staat)	DE
II-7	Sitz/Wohnsitz (Staat)	DE
II-8	Telefonnr.	(089) 636-82819
II-9	Telefaxnr.	(089) 636-81857
III-1	**Anmelder und/oder Erfinder**	
III-1-1	Diese Person ist	Anmelder und Erfinder
III-1-2	Anmelder für	Nur US
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III-1-6	Staatsangehörigkeit (Staat)	IT
III-1-7	Sitz/Wohnsitz (Staat)	DE

IV-1	**Anwalt oder gemeinsamer Vertreter; oder besondere Zustellanschrift** Die unten bezeichnete Person ist/wird hiermit bestellt, um den (die) Anmelder vor den internationalen Behörden zu vertreten, und zwar als:	gemeinsamer Vertreter
IV-1-1	Name	SIEMENS AKTIENGESELLSCHAFT
IV-1-2	Anschrift:	Postfach 22 16 34 D-80506 München Deutschland
IV-1-3	Telefonnr.	(089) 636-82819
IV-1-4	Telefaxnr.	(089) 636-81857
V	**Bestimmung von Staaten**	
V-1	Regionales Patent (andere Schutzrechtsarten oder Verfahren sind ggf. in Klammern nach der (den) betreffenden Bestimmung(en) angegeben)	EP: AT BE CH&LI CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE und jeder weitere Staat, der Mitgliedsstaat des Europäischen Patentübereinkommens und Vertragsstaat des PCT ist
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V-6	**Staaten, die von der Erklärung über vorsorgliche Bestimmungen ausgenommen werden**	KEINE
VI-1	**Priorität einer früheren nationalen Anmeldung beansprucht**	
VI-1-1	Anmeldedatum	27 Oktober 1998 (27.10.1998)
VI-1-2	Aktenzeichen	19849549.8
VI-1-3	Staat	DE
VI-2	**Ersuchen um Erstellung eines Prioritätsbeleges** Das Anmeldeamt wird ersucht, eine beglaubigte Abschrift der in der (den) nachstehend genannten Zeile(n) bezeichneten früheren Anmeldung(en) zu erstellen und dem internationalen Büro zu übermitteln:	VI-1

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VII-1	**Gewählte Internationale Recherchenbehörde**	Europäisches Patentamt (EPA) (ISA/EP)	
VIII	**Kontrolliste**	Anzahl der Blätter	Elektronische Datei(en) beigefügt
VIII-1	Antrag	4	-
VIII-2	Beschreibung	14	-
VIII-3	Ansprüche	4	-
VIII-4	Zusammenfassung	1	98p2958.txt
VIII-5	Zeichnung(en)	1	-
VIII-7	INSGESAMT	24	
	Beigefügte Unterlagen	Unterlage(n) in Papierform beigefügt	Elektronische Datei(en) beigefügt
VIII-8	Blatt für die Gebührenberechnung	✓	-
VIII-16	PCT-EASY-Diskette	-	Diskette
VIII-17	Sonstige (einzeln aufgeführt):	Kopie der Ursprungsfassung	-
VIII-18	**Nr. der Abb. der Zeichn., die mit der Zusammenf. veröffentlicht werden soll**	1	
VIII-19	**Sprache der int. Anmeldung**	Deutsch	
IX-1	**Unterschrift des Anmelders oder Anwalts**	i. V. Margraf	
IX-1-1	Name	SIEMENS AKTIENGESELLSCHAFT	
IX-1-2	Name der unterzeichnenden Person	Margraf	
IX-1-3	Eigenschaft	Nr. 144/74 Ang.AV	
IX-2	**Unterschrift des Anmelders oder Anwalts**	Obradovic D.	
IX-2-1	Name (FAMILIENNAME, Vorname)	OBRADOVIC, Dragan	

VOM ANMELDEAMT AUSZUFÜLLEN

10-1	**Datum des tatsächlichen Eingangs dieser internationalen Anmeldung**	
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2	**Anmelder**	SIEMENS AKTIENGESELLSCHAFT, et al.

			Höhe der Gebühr/Multiplikator	Gesamtbeträge (DEM)
12	**Berechnung der vorgeschriebenen Gebühren**			
12-1	Übermittlungsgebühr	T	⇨	150
12-2	Recherchengebühr	S	⇨	1.848,26
12-3	Internationale Gebühr Grundgebühr (erste 30 Blätter)	b1	807,76	
12-4	Anzahl der Blätter über 30		0	
12-5	Zusatzblattgebühr	(X)	19,56	
12-6	Gesamtbetrag der weiteren Gebühren	b2	0	
12-7	b1 + b2 =	B	807,76	
12-8	Bestimmungsgebühren Anzahl der in der internationalen Anmeldung vorgenommenen Bestimmungen		3	
12-9	Anzahl der zu zahlenden Bestimmungsgebühren (höchstens 10)		3	
12-10	Bestimmungsgebühr	(X)	185,8	
12-11	Gesamtbetrag der Bestimmungsgebühren	D	557,4	
12-12	PCT-EASY-Gebührenermäßigung	R	-248,39	
12-13	Gesamtbetrag der internationalen Gebühr (B+D+R)	I	⇨	1.116,77
12-14	Gebühr für Prioritätsbeleg Anzahl der beantragten Prioritätsbelege		1	
12-15	Gebühr per Prioritätsbeleg	(X)	35	
12-16	Gesamtbetrag Gebühr für Prioritätsbeleg(e)	P	⇨	35
12-17	**GESAMTBETRAG DER ZU ZAHLENDEN GEBÜHREN (T+S+I+P)**		⇨	3.150,03
12-19	**Zahlungsart**		Sonstige: Abbuchung durch gesonderte Zahlungsliste	

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12-21	Nummer des laufenden Kontos	409022601
12-22	Datum	08 Oktober 1999 (08.10.1999)
12-23	Name und Unterschrift	SIEMENS AKTIENGESELLSCHAFT i. V.

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13-2-2	**Prüfergebnisse** **Staaten**	Grün? Es können mehr Bestimmungen vorgenommen werden. Bitte überprüfen.

Beschreibung

Verfahren und Anordnung zur Ermittlung von Parametern eines technischen Systems

Die Erfindung betrifft ein Verfahren sowie einer Anordnung zur Ermittlung von Parametern eines technischen Systems

Bei einer Vielkanal-Übertragung und einem Vielkanal-Empfang von Signalen tritt häufig eine Interferenz beispielsweise zwischen den Signalen/Bildern auf. Ein typisches Beispiel ist dabei eine Mischung eines Sprachsignals mit Rauschen, was bei der Telekommunikation und bei Videokonferenzen ein großes Problem darstellen kann. Die vorliegende Erfindung betrifft daher das Gebiet der Signaltrennung, um beispielsweise ein ursprüngliches Sprachsignal wiederzugewinnen.

Typische bekannte Techniken zur Trennung der Quellensignale ausgehend von Mischsignalen basieren auf einer zeitlichen Mittelung oder einer Filterung der Signale. Dies hat indessen Nachteile bezüglich des Rechenaufwands zur Folge.

Es sind auch Verfahren auf der Grundlage der sogenannten Blind Channel Equalization (Signalentzerrung ohne Vorkenntnisse über den Übertragungskanal) bekannt, aber diese Verfahren benötigen immer eine gewisse Kenntnis über die Quellsignale, wie beispielsweise eine Kenntnis über ihre statistische Verteilung.

Das Problem der Signaltrennung tritt beispielsweise auch auf, wenn zwei Sprecher in zwei entfernt von ihnen stehende Mikrofone sprechen, so daß jedes Mikrofon ein Gemisch der von den zwei Sprechern gesprochenen Signale aufnimmt. Somit besteht das Problem, das Signalgemisch, also eine Menge überlagerter Eingangssignale wieder zu trennen. Aus L.Molgedey, H.G. Schuster, „Separation of a Mixture of Independent Signals using Time-Delayed Correlations", Phys. Ref. Lett. 72, 3634 (1994) ist dabei das folgende Verfahren bekannt: Das Problem, n

überlagerte und korrelierte Quellensignale (Eingangssignale) zu trennen und gleichzeitige die Mischungskoeffizienten der Quellenstärken zu bestimmen, läßt sich auf ein Eigenwertproblem reduzieren, bei dem simultan zwei symmetrische $n \times n$ Matrizen diagonalisiert werden müssen. Die Matrixelemente sind meßbare zeitverzögerte Korrelationsfunktionen. Die Lösung des Eigenwertproblems kann durch ein neuronales Netz erfolgen, wobei die Lernregel für die lateralen inhibitorischen Wechselwirkungen zwischen den Neuronen durch eine Liapunov-Funktion bestimmt werden kann, deren Minima die (entarteten) Lösungen des Problems liefern.

Dieses Verfahren wurde auch bereits auf akustische Eingangssignale angewandt (siehe F. Ehlers, H. G. Schuster, „Blind Separation of convolutive mixtures and an application in automatic speech recognition" IEEE Trans. Signal Proc. (1997).

Aus der DE 195 31 388 C1 ist ein Signaltrennungsverfahren und eine Signaltrennungseinrichtung für nichtlineare Mischungen unbekannter Signale (Blind Channel) bekannt, das schematisch in Fig. 3 dargestellt ist.

Dieses deutsche Patent behandelt die Separierung eines Signalgemisches, bestehend aus der nichtlinearen Überlagerung von M unbekannten Quellsignalen X1, X2, wobei N ($N \geq M$) unterschiedliche Mischungen der M Quellsignale X1, X2 inklusive eines eventuellen Störsignals einer Signalauswerteeinrichtung zugeführt werden, die durch eine statistische Analyse der Signale die nichtlinearen Übertragungsfaktoren bestimmt und mit diesen errechneten Faktoren die Mischung rückgängig macht, so daß die N Ausgänge der Signaltrennungseinrichtung möglichst näherungsweise die M Quellsignale ohne Überlagerungen enthalten. Dadurch wird eine Behandlung nichtlinearer Gemische möglich, wobei nichtlinear bedeutet, daß die Quellsignale X1, X2 durch ein unbekanntes nichtlineares System G gemischt werden. Das unbekannte System G wird durch eine sogenannte Volterra-Reihe beschrieben, und die Signaltrennungseinrichtung G-1 bestimmt die Koeffizienten der Volterra-Reihe. Mit deren

Kenntnis ist eine Entmischung des Signalgemischs möglich. Außerdem können die Koeffizienten zu weiteren Analysen zur Orts- oder Geschwindigkeitsbestimmung der Signalquellen benutzt werden.

Das aus dieser Druckschrift bekannte Verfahren besteht dabei im wesentlichen aus zwei Schritten:

- Erstens werden die durch den wählbaren Grad der Nichtlinearität bei der Mischung eindeutig bestimmten nichtlinearen Gleichungen für ein gleitendes Zeitfenster gelöst und die Lösungen werden über die Zeit gemittelt. Diese zeitliche Mittelung stellt einen Hauptnachteil dieser bekannten Technik dar, da sie den Rechenaufwand und gleichzeitig die Zeitdauer für die Berechnung erhöht.

- Zweitens wird aus einer genügend großen Anzahl von unterschiedlichen Kumulanten der geschätzten Ausgangssignale gebildetes Potential minimiert, wobei die zur Berechnung des Potentials nötigen Werte aus einem gleitenden Zeitfenster einer wählbaren Länge stammen. Dabei ist vorausgesetzt, daß sich das Mischungssystem so langsam ändert, daß diese Änderung bei der Berechnung der gesuchten Mischungsfaktoren vernachlässigt werden kann. Gemäß diesem deutschen Patent wird zur Ausführung des zweiten genannten Schrittes eine Kostenfunktion konstruiert, die minimiert wird. Wenn das globale Minimum erreicht ist, hat man die optimalen Werte, in diesem Fall der Übertragungsfaktoren, gefunden.

Hinsichtlich des zeitlichen Aufwands sowie des Rechenaufwands ist das in der DE 195 31 388 C1 beschriebene Verfahren aufgrund der Ausführung einer zeitlichen Mittelung am Abschluß des ersten, oben genannten Verfahrensschrittes nachteilig.

Die vorliegende Erfindung hat daher zur Aufgabe, ein Verfahren und eine Anordnung bereitzustellen, die die Trennung überlagerter, statistisch voneinander unabhängiger akustischer Signale mit verringertem Rechenaufwand ermöglicht.

Diese Aufgabe wird durch das Verfahren sowie durch die Anordnung mit den Merkmalen gemäß den unabhängigen Ansprüchen gelöst.

Bei einem Verfahren zur Ermittlung von Parametern eines technischen Systems, mit dem Ausgangssignale aus einer Menge überlagerter, statistisch voneinander unabhängiger Eingangssignale ermittelt werden können, werden die Parameter derart ermittelt, daß die statistische Unabhängigkeit der Ausgangssignale maximiert wird.

Eine Anordnung zur Ermittlung von Parametern eines technischen Systems, mit dem Ausgangssignale aus einer Menge überlagerter, statistisch voneinander unabhängiger Eingangssignale ermittelt werden können, weist einen Prozessor auf, der derart eingerichtet ist, daß die Parameter derart ermittelt werden können, daß die statistische Unabhängigkeit der Ausgangssignale maximiert wird.

Vorteilhafte Weiterbildungen der Erfindung ergeben sich aus den abhängigen Ansprüchen.

Die Parameter werden bevorzugt in einem iterativen Verfahren ermittelt.

In einer weiteren Ausgestaltung sind die Parameter Elemente einer Entmischmatrix, mit der die Menge der überlagerten Eingangssignale multipliziert oder auch gefaltet wird, wodurch die Ausgangssignale gebildet werden.

Die Optimierung der Parameter der Entmischmatrix wird bevorzugt durch die folgenden Schritte erhalten:

- Wiederholung einer zeitverzögerten Dekorrelationsberechnung zur Ermittlung der Eigenwerte der Entmischmatrix,

- Ermittlung der Eigenwerte der Entmischmatrix, für die Kreuzkorrelationen einen minimalen Wert annehmen, und
- Ausführung einer Kumulantenminimierung, wobei als Startwerte für die Kumulantenminimierung die im vorherigen Schritt ermittelten Eigenwerte verwendet werden.

Die Kumulantenminimierung kann beispielsweise durch Trainieren eines neuronalen Netzes, aber auch jegliche andere bekannte Minimierungstechnik, wie beispielsweise Gradientenabstieg oder Monte-Carlo-Simulationen verwendet werden.

In einer Weiterbildung wird bei der Optimierung der Parameter der Entmischmatrix wenigstens ein Diagonalparameter der Entmischmatrix auf einen vorgegebenen Wert gesetzt, womit , um eine Stabilität des Minimierungsvorgangs hin zu einem globalen Minimum zu gewährleisten.

Die Entmischmatrix wird bevorzugt auf eine finite Impulsantwort begrenzt, d.h. es wird ein FIR-Filter (Finite Impulse Response) eingesetzt zur Bildung der einzelnen Komponenten der Entmischmatrix. Der FIR-Filter kann sowohl ein kausales FIR-Filter oder auch ein nicht-kausales FIR-Filter sein.

Ferner wird die Entmischmatrix während der Kumulantenminimierung bevorzugt durch Projektion in einen Einheitskreis stabilisiert.

Die Weiterbildungen gelten sowohl für das Verfahren als auch für die Anordnung, wobei bei der Anordnung jeweils der Prozessor derart eingerichtet ist, daß der entsprechende Verfahrensschritt durchführbar ist oder durchgeführt wird.

Die Erfindung sowie deren Weiterbildungen können vorteilhaft eingesetzt werden zur Trennung überlagerter, statistisch voneinander unabhängiger Eingangssignale, insbesondere akustischer Eingangssignale.

Das Verfahren sowie die Anordnung sind für eine beliebige Anzahl von Eingangssignalen anwendbar.

Weitere Vorteile, Merkmale und Eigenschaften der vorliegenden Erfindung werden nunmehr bezugnehmend auf die beiliegenden Figuren der Zeichnungen anhand eines Ausführungsbeispiels näher erläutert.

Fig. 1 zeigt die Anwendung eines Systems zur Trennung überlagerter, statistisch voneinander unabhängiger akustischer Signale gemäß dem Ausführungsbeispiel,

Fig. 2 zeigt eine symbolische Darstellung des Systems von Fig. 1, und

Fig. 3 zeigt eine aus dem Stand der Technik (DE 195 31 388 C1) bekannte Signaltrennungseinrichtung für nichtlineare Mischungen unbekannter Signale.

Zur Wiedergewinnung des ursprünglichen Sprachsignals wird aus einer Mischung von Signalen die statistische Unabhängigkeit zwischen den Quellensignalen (ursprüngliches Sprachsignal und Rauschen), im weiteren auch als Eingangssignale bezeichnet, ausgenutzt und der inverse Vorgang des dynamischen Systems, der die Mischung der Signale ergeben hat, wird im wesentlichen näherungsweise trainiert (gelernt). Zwei verschiedene Mischungen des Sprachsignals bzw. des Rauschsignals werden beispielsweise durch zwei Mikrofone 1, 2 (vgl. **Fig.1**) erhalten, die voneinander beabstandet sind und/oder in entgegengesetzten Richtungen ausgerichtet sind. Bei dem Verfahren wird die sogenannte zeitverzögerte Dekorrelationstechnik (TDD, time delayed decorrelation) verwendet, um die Lernphase zu initiieren, d.h. um Startwerte für die Lernphase zu ermitteln und vorzugeben, wodurch der Berechnungsaufwand für eine im weiteren beschriebenen Kumulantenminimierung verringert werden kann und die Gefahr lokaler Minima verringert werden kann.

Fig.1 zeigt zwei Mikrofone 1, 2, die ein erstes Eingangssignal Z1(t) und ein zweites Eingangssignal Z2(t) aufnehmen. Diese Eingangssignale Z1(t) und Z2(t) können untereinander wiederum jeweils mit Rauschen vermischt sein, was durch eine Mischmatrix S (siehe Bezugszeichen 3) symbolisch in **Fig.1** dargestellt ist. Nach dem Empfang bzw. der Übertragung wird eine Menge X1(t) und X2(t) überlagerter, statistisch voneinander unabhängiger Eingangssignale Z1(t) und Z2(t) erhalten. Diese Signale werden in eine Berechnungseinheit 4 eingegeben, in der im wesentlichen zwei Schritte ausgeführt werden, die symbolisch durch eine Berechnungseinheit B (Bezugszeichen 6) für den ersten Schritt sowie ein neuronales Netzwerk 5 für den zweiten Schritt dargestellt ist.

Durch die Berechnungseinheit 4 werden zwei Ausgangssignale Y1(t) bzw. Y2(t) ermittelt, die bei optimaler Einstelllung der Parameter in der Berechnungseinheit 4 näherungsweise gleich den Eingangssignalen Z1(t) bzw. Z2(t) sind. Mit anderen Worten, bei optimaler Einstellung der Parameter der verwendeten Matrizen in der Berechnungseinheit 4 erfolgt durch diese Berechnungseinheit 4 im wesentlichen der inverse Vorgang zu dem dynamischen Mischvorgang, der durch die Matrix S (Bezugszeichen 3) symbolisch dargestellt ist. Das Ausführungsbeispiel beschäftigt sich mit dem Optimierungsvorgang der Einstellung der Parameter der Entmischmatrix.

Es werden die Parameter der Matrizen in der Berechnungseinheit 4 derart optimiert, daß die statistische Unabhängigkeit zwischen den durch den Matrizenvorgang in der Berechnungseinheit 4 gewonnenen Ausgangssignale Y1(t), Y2(t) maximiert wird. Zu diesem Zweck werden die Ausgangssignale Y1(t) bzw. Y2(t) zu der Berechnungseinheit 4 zurückgeführt (s. Rückführschleifen 7 bzw. 8). Durch ein iteratives Verfahren wird ermittelt, ob sich die statistische Unabhängigkeit der Ausgangssignale Y1(t) bzw. Y2(t) im Vergleich zu dem vorherigen Schritt der Iteration erhöht hat (und somit die Iteration die „richtige" Richtung in Richtung des globalen Minimums einer im weiteren beschriebenen Kostenfunktion einnimmt) oder nicht.

Fig.2 zeigt eine mathematische Darstellung des Schemas von **Fig.1**, wobei der Mischvorgang 3 durch eine Matrix S(q) mathematisch beschrieben werden kann und der Entmischvorgang, der durch die Berechnungseinheit 4 erfolgen soll, durch eine Entmischmatrix M(q) symbolisiert wird.

In **Fig.2** ist somit das Problem der Trennung einer sogenannten Multi-Channel Blind Source (Vielfach-Kanal-Quelle ohne a-priori-Kenntnis) in zwei Dimensionen dargestellt. Dabei ist angenommen, daß das Mischsystem S(q), wobei q für eine Einheitsverzögerung steht, stabil ist und gleichzeitig auch eine stabile Invertierung aufweist, d.h. daß es ein Minimalphasensystem ist. Darüber hinaus ist angenommen, daß die Eingangssignale Z1(t) und Z2(t) (beispielsweise ein Sprach- bzw. ein Rauschsignal) statistisch voneinander unabhängig sind und nicht gaussförmig verteilt sind. Die Menge X1(t) und X2(t) der überlagerten Eingangssignale Z1(t) und Z2(t) sind Eingangssignale in ein Entmischsystem mit einer Entmischmatrix M(q), deren Parameter (Matrixelemente) auf eine Maximierung der statistischen Unabhängigkeit zwischen den Ausgangssignalen Y1(t) und Y2(t) trainiert werden. Unter „Trainieren" ist dabei der gut bekannte Lernvorgang beispielsweise eines neuronalen Netzes bekannt, das als ein Beispiel für eine Technik genannt sein soll, die statistische Unabhängigkeit zu maximieren. Dies erfolgt durch Minimierung einer im weiteren beschriebenen Kostenfunktion J(M).

Es wird eine Kumulanten-Kostenfunktion gebildet, die die Diagonalkumulantenelementen der Kumulantenordnung 2 - 4 minimiert:

$$\mathrm{Dcum} \approx J(M) = \sum_{i=1}^{4} \sum_{\mathrm{nondiag}} \left[C^{(i)}\mathrm{nondiag}\right]^2$$

Folgende Gesichtspunkte der dynamischen Mischung durch die Mischmatrix S(q) sind dabei zu berücksichtigen:

- Stabilität des Entmischsystems:
  Dies wird erreicht, wenn M(q) auf eine finite Impulsantantwort (FIR-Filter) beschränkt wird. Die Stabilität des FIR-Systems M(q) kann darüber hinaus auch dadurch erhalten werden, daß während der Lernphase eine Projektion in den Einheitskreis erfolgt Eine möglicherweise vorliegende Nichtkausalität der Invertierung von S(q) kann durch eine geeignete Zeitverschiebung (Verzögerung) des Eingangssignals X(t) kompensiert werden.

- Eindeutigkeit der getrennten Signale Y(t):
  Für den Fall statischer Mischungen werden die ursprünglichen Quellsignale durch eine Skalierung wiedergewonnen. Für den Fall einer dynamischen Entmischung ist die Gefahr einer Nichteindeutigkeit der getrennten Signale Y(t) sogar noch größer. Es ist offensichtlich, daß für den Fall, daß Y1(t) und Y2(t) statistisch voneinander unabhängig sind, auch jegliche linear gefilterte Modifikation dieser Signale immer noch statistisch unabhängig sind. Daher ist eine zusätzliche Information notwendig, um die inhärente Nichteindeutigkeit des Problems zu verringern.

- Gaussverformung der Daten:
  Algorithmen auf Kumulantenbasis für eine statische Blind Source-Trennung minimieren bzw. eliminieren in effektiver Weise Diagonalkumulanten höherer Ordnung entsprechend den Ausgangssignalen Y(t). Andererseits führt eine lineare Filterung zu einer Gaussverteilungs-Verformung der Daten, bei denen die Kumulanten höherer Ordnung in Richtung 0 gehen. Dies kann daher zu Randlösungen führen, in denen die Kostenfunktion ein lokales Minimum erreicht, wobei die erwünschte eigentliche Trennung (globales Minimum) nicht erfolgt. Um diesen ungewünschten Fall zu vermeiden, wird die Struktur der Entmisch-Transferfunktion (Entmischmatrix) M(q) einigen Beschränkungen unterworfen.

Um die obengenannten Probleme zu umgehen, wird ein Ansatz gewählt, daß wenigstens eines (oder auch alle) der Diagonalelemente auf den Einheitswert gesetzt werden:

M11(q) = 1

und

M22(q) = 1.

Diese Annahme ist exakt, wenn die Mischelemente S11(q) und/oder S22(q) ebenfalls einen Einheitswert „1" aufweisen. Sonst sei angenommen, daß S11(q) und/oder S22(q) stabile Invertierungen aufweisen, was die Skalierung der Diagonalelemente von M(q) auf den Einheitswert erlaubt. Dieser Ansatz verringert wesentlich die Nichteindeutigkeit einer Lösung und vermeidet darüber hinaus in effektiver Weise die Gefahr einer übermäßigen Gaussverteilungs-Verformung der Ausgangssignale. Auch wenn auf den ersten Blick die Beschränkung der Diagonalelemente von M(q), wie oben ausgeführt, sehr restriktiv wirkt, ist diese Annahme in der praktischen Anwendung in der Regel erfüllt. Ein typisches Beispiel ist die Entrauschung von Sprachsignalen auf der Grundlage einer Aufzeichnung mit zwei Mikrofonen, wobei die Mikrofone voneinander räumlich getrennt sind oder ein Mikrofon in Richtung des Sprechers gerichtet ist, während das andere Mikrofon in der umgekehrten Richtung gerichtet ist, so daß das zweite, von dem Sprecher abgewandte Signal im wesentlichen nur ein Rauschsignal aufnimmt.

Der Kumulantenansatz basiert auf einer direkten Ermittlung der Diagonalkumulanten, wie sie in dem eingangs genannten Artikel von F. Ehlers und H. Schuster, „Blind Separation of convolutive mixtures and an application in automatic speech recognition" IEEE Trans. Signal Proc. (1997), ausgeführt ist, was aber inhärent den Nachteil aufweist, daß die numerische Lösung sehr aufwendig ist. Daher wird eine geeignete Initialisierung dieses Minimierungsverfahrens verwendet. Um Startwerte für das

Minimierungsverfahren zu ermitteln, wird gemäß der vorliegenden Erfindung die Technik der zeitverzögerten Dekorrelation (TDD) zur gleichzeitigen Dekorrelierung zweier verschiedener Zeitverzögerungen angewendet, wobei diese TDD-Technik sich auf ein geeignetes Matrix-Eigenwertproblem zurückführen läßt. Wie bereits gesagt, wird diese TDD-Technik gemäß der vorliegenden Erfindung zur Initiierung des Diagonal(Kreuzkorrelations)-Kumulantenminimierungsproblems verwendet.

Zusammenfassend läßt sich das Verfahren in die folgenden beiden Schritte unterteilen:

1. Wiederholung des TDD-Verfahrens auf Grundlage des Eigenwertproblems im Frequenzbereich für verschiedene Verzögerungspaare und Ermittlung der Lösung, bei der die Kreuzkorrelationsterme einen minimalen Wert aufweisen.

2. Initiierung (Start) der Diagonal-Kumulantenminimierung auf Grundlage der im oben genannten Schritt ermittelten Startwerte (FIR-Parameter).

Im folgenden sollen noch einige Haupteigenschaften und Vorteile noch einmal zusammengefaßt werden:

- Es ist kein a-priori-Wissen der Signaleigenschaften notwendig, mit der Ausnahme, daß eine statistische Unabhängigkeit gefordert wird.

- Die Stabilität des dynamischen Entmischsystems wird durch die Modulierung seiner Komponenten als FIR-Filter gewährleistet.

- Eine übermäßige Gaussverteilung-Verformung wird durch den Ansatz vermieden, daß wenigstens eines der Elemente der Mischtransfer-Funktionsmatrix (Entmischmatrix) auf den Einheitswert gesetzt wird bzw. auf den Einheitswert skaliert werden kann, und

- da der Kumulanten-Minimierungsschritt (Schritt 2) einen hohen Rechenaufwand erfordert, wird der Lernalgorithmus beispielsweise eines neuronalen Netzes durch das TDD-Verfahren initialisiert.

Im weiteren ist ein Programm in Matlab, Version 4 oder Version angegeben, mit dem das oben beschriebene Ausführungsbeispiel auf einem Rechner realisiert werden kann:

```
function[cost,out1,out2]=cumulant_costFIRa2(par,input,p1,p11,
p2,p22,a3,a4);
%[cost,out1,out2]=cumulant_costFIRa2(par,input,p1,p11,p2,p22,
a3,a4);
% cumulant cost
% FIR representation used
% filter function used in both directions (non_causal)

[np,mp]=size(par);
fir1=par(1:p1);
fir11=par(1+p1:p1+p11);
fir2=par(1+p1+p11:p1+p11+p2);
fir22=par(p1+p11+p2+1:mp);
den=1;                                  %FIR only

out1=[input(:;1)-filter(fir1,den,input(:,2))-flipud(filter
([0 fir11 ],[den],flipud(input(:,2))))];
                        %/std(input(1,:));      %dlsim

%filter
out2=[input(:,2)-filter(fir2,den,input(:,1))-flipud(filter
([0 fir22 ],[den],flipud(input(:,1))))];%/std(input(:,2));
                        %dlsim    %filter

out=[out1 out2];
%out1=out1/std(out1); % this scaling was not needed in examples
in SIP98 paper
%out2=out2/std(out2);

Ld=0;  % number of delays in calculating the cross-correlation
cost3=0;
cost4=0;
```

```
costALL1=[];
costALL2=[];
o12=out1.*out2;
cost2=mean(o12)^2;

o112=out1.*out1.*out2;
o122=out1.*out2.*out2;

if a3 ==1
     cost3=[mean(o122)]^2+[mean(o122)]^2;
end

if a4 ==1

     cost4=[mean(o112.*out1)-3*mean(out1.^2)*mean(o12)]^2+...
[mean((out1.^2).*(out2.^2))-2*mean(o12)^2-
mean(out1.^2)*mean(out2.^2)]^2+...
[mean(o122.*out2)-3*mean(out2.^2)*mean(o12)]^2;

end

%cum4a=[cum4x(out1,out1,out1,out1)]^2;
%cum4b=[cum4x(out2,out2,out2,out2)]^2;

cost=cost2+a3*cost3+a4*cost4;          %-cum4a-cum4b;
```

Patentansprüche

1. Verfahren zur Ermittlung von Parametern eines technischen Systems, mit dem Ausgangssignale aus einer Menge überlagerter, statistisch voneinander unabhängiger Eingangssignale ermittelt werden können, bei dem die Parameter derart ermittelt werden, daß die statistische Unabhängigkeit der Ausgangssignale maximiert wird.

2. Verfahren nach Anspruch 1,
bei dem die Parameter in einem iterativen Verfahren ermittelt werden.

3. Verfahren nach Anspruch 1 oder 2,
bei dem die Parameter Elemente einer Entmischmatrix sind, mit der die Menge der überlagerten Eingangssignale multipliziert wird, wodurch die Ausgangssignale gebildet werden.

4. Verfahren nach Anspruch 3,
bei dem die Optimierung der Parameter der Entmischmatrix durch die folgenden Schritte erhalten wird:
- Wiederholung einer zeitverzögerten Dekorrelationsberechnung (6) zur Ermittlung der Eigenwerte der Entmischmatrix,
- Ermittlung der Eigenwerte der Entmischmatrix, für die Kreuzkorrelationen einen minimalen Wert annehmen, und
- Ausführung einer Kumulantenminimierung (5), wobei als Startwerte für die Kumulantenminimierung die im vorherigen Schritt ermittelten Eigenwerte verwendet werden.

5. Verfahren nach Anspruch 4,
bei dem die Kumulantenminimierung durch Trainieren eines neuronalen Netzes (5) erfolgt.

6. Verfahren nach einem der Ansprüche 3 bis 5,

bei dem bei der Optimierung der Parameter der Entmischmatrix wenigstens ein Diagonalparameter der Entmischmatrix auf einen vorgegebenen Wert gesetzt wird.

7. Verfahren nach einem der Ansprüche 3 bis 6,
bei dem die Entmischmatrix auf eine finite Impulsantwort begrenzt wird.

8. Verfahren nach einem der Ansprüche 3 bis 7,
bei dem die Entmischmatrix während der Kumulantenminimierung (5) durch Projektion in einen Einheitskreis stabilisiert wird.

9. Verfahren nach einem der Ansprüche 1 bis 8, eingesetzt zur Trennung überlagerter, statistisch voneinander unabhängiger Eingangssignale.

10. Verfahren nach einem der Ansprüche 1 bis 8, eingesetzt zur Trennung überlagerter, statistisch voneinander unabhängiger akustischer Eingangssignale.

11. Anordnung zur Ermittlung von Parametern eines technischen Systems, mit dem Ausgangssignale aus einer Menge überlagerter, statistisch voneinander unabhängiger Eingangssignale ermittelt werden können, mit einem Prozessor, der derart eingerichtet ist, daß die Parameter derart ermittelt werden können, daß die statistische Unabhängigkeit der Ausgangssignale maximiert wird.

12. Anordnung nach Anspruch 11,
bei der der Prozessor derart eingerichtet ist, daß die Parameter in einem iterativen Verfahren ermittelt werden.

13. Anordnung nach Anspruch 11 oder 12,
bei der der Prozessor derart eingerichtet ist, daß die Parameter Elemente einer Entmischmatrix sind, mit der die Menge der überlagerten Eingangssignale multipliziert wird, wodurch die Ausgangssignale gebildet werden.

14. Anordnung nach Anspruch 13, bei der der Prozessor derart eingerichtet ist, daß die Optimierung der Parameter der Entmischmatrix durch die folgenden Schritte erhalten wird:
- Wiederholung einer zeitverzögerten Dekorrelationsberechnung (6) zur Ermittlung der Eigenwerte der Entmischmatrix,
- Ermittlung der Eigenwerte der Entmischmatrix, für die Kreuzkorrelationen einen minimalen Wert annehmen, und
- Ausführung einer Kumulantenminimierung (5), wobei als Startwerte für die Kumulantenminimierung die im vorherigen Schritt ermittelten Eigenwerte verwendet werden.

15. Anordnung nach Anspruch 14, bei der der Prozessor derart eingerichtet ist, daß die Kumulantenminimierung durch Trainieren eines neuronalen Netzes (5) erfolgt.

16. Anordnung nach einem der Ansprüche 13 bis 15, bei der der Prozessor derart eingerichtet ist, daß bei der Optimierung der Parameter der Entmischmatrix wenigstens ein Diagonalparameter der Entmischmatrix auf einen vorgegebenen Wert gesetzt wird.

17. Anordnung nach einem der Ansprüche 13 bis 16, bei der der Prozessor derart eingerichtet ist, daß die Entmischmatrix auf eine finite Impulsantwort begrenzt wird.

18. Anordnung nach einem der Ansprüche 13 bis 17, bei der der Prozessor derart eingerichtet ist, daß die Entmischmatrix während der Kumulantenminimierung (5) durch Projektion in einen Einheitskreis stabilisiert wird.

19. Anordnung nach einem der Ansprüche 11 bis 18, eingesetzt zur Trennung überlagerter, statistisch voneinander unabhängiger Eingangssignale.

20. Anordnung nach einem der Ansprüche 11 bis 18, eingesetzt zur Trennung überlagerter, statistisch voneinander unabhängiger akustischer Eingangssignale.

Zusammenfassung

Verfahren und Anordnung zur Ermittlung von Parametern eines technischen Systems

Es werden Parameter eines technischen Systems, mit dem Ausgangssignale aus einer Menge überlagerter, statistisch voneinander unabhängiger Eingangssignale ermittelt werden können, bestimmt. Die Parameter werden derart ermittelt, daß die statistische Unabhängigkeit der Ausgangssignale maximiert wird.

Sign. Fig. 1

FIG 1

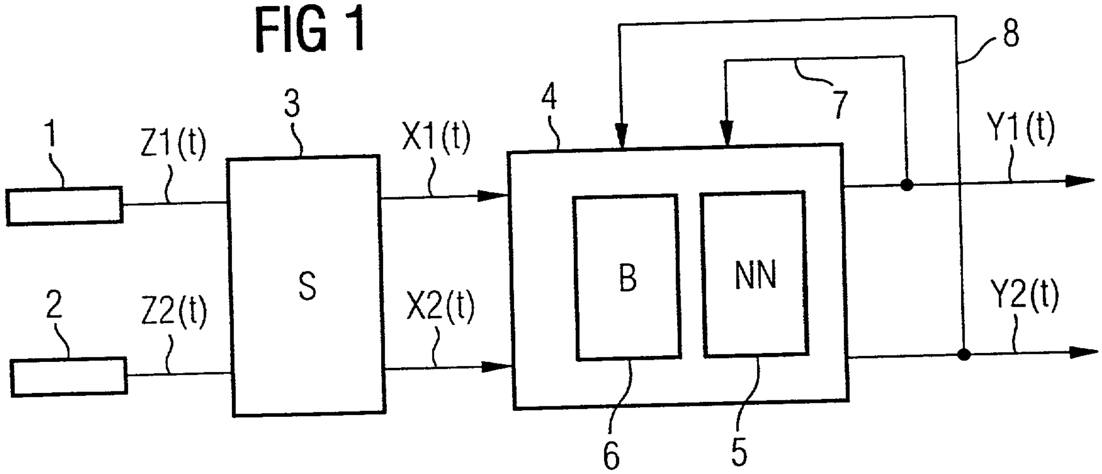


FIG 2

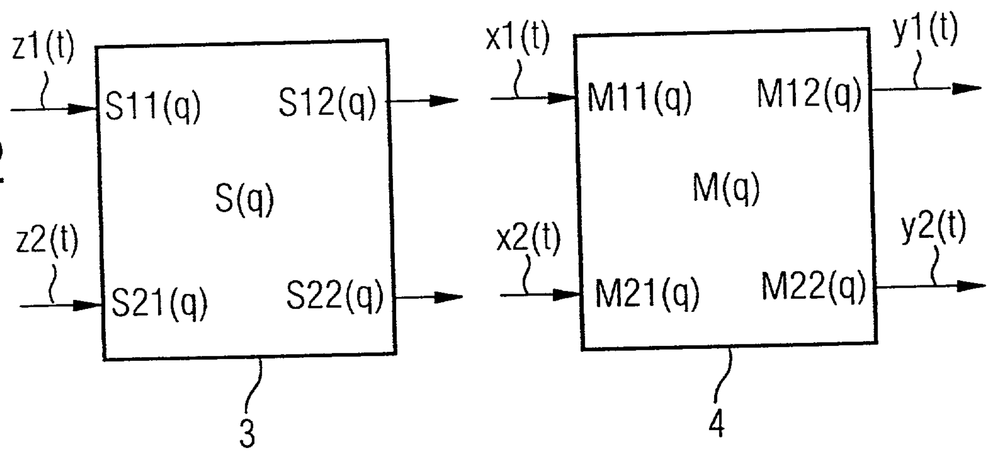
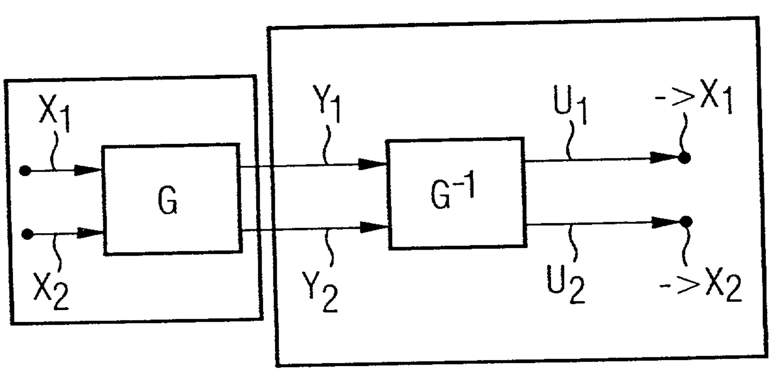


FIG 3



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Erklärung Für Patentanmeldungen Mit Vollmacht
German Language Declaration

Als nachstehend benannter Erfinder erkläre ich hiermit an Eides Statt:

As a below named inventor, I hereby declare that:

dass mein Wohnsitz, meine Postanschrift, und meine Staatsangehörigkeit den im Nachstehenden nach meinem Namen aufgeführten Angaben entsprechen,

My residence, post office address and citizenship are as stated below next to my name,

dass ich, nach bestem Wissen der ursprüngliche, erste und alleinige Erfinder (falls nachstehend nur ein Name angegeben ist) oder ein ursprünglicher, erster und Miterfinder (falls nachstehend mehrere Namen aufgeführt sind) des Gegenstandes bin, für den dieser Antrag gestellt wird und für den ein Patent beantragt wird für die Erfindung mit dem Titel:

I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled

Signaltrennungsverfahren und -anordnung für nichtlineare Mischungen unbekannter Signale

Signal separation method and device for non-linear mixing of unknown signals

deren Beschreibung

(zutreffendes ankreuzen)
☐ hier beigefügt ist.
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Ich bestätige hiermit, dass ich den Inhalt der obigen Patentanmeldung einschliesslich der Ansprüche durchgesehen und verstanden habe, die eventuell durch einen Zusatzantrag wie oben erwähnt abgeändert wurde.

I hereby state that I have reviewed and understand the contents of the above identified specification, including the claims as amended by any amendment referred to above.

Ich erkenne meine Pflicht zur Offenbarung irgendwelcher Informationen, die für die Prüfung der vorliegenden Anmeldung in Einklang mit Absatz 37, Bundesgesetzbuch, Paragraph 1.56(a) von Wichtigkeit sind, an.

I acknowledge the duty to disclose information which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, §1.56(a).

Ich beanspruche hiermit ausländische Prioritätsvorteile gemäss Abschnitt 35 der Zivilprozessordnung der Vereinigten Staaten, Paragraph 119 aller unten angegebenen Auslandsanmeldungen für ein Patent oder eine Erfindersurkunde, und habe auch alle Auslandsanmeldungen für ein Patent oder eine Erfindersurkunde nachstehend gekennzeichnet, die ein Anmeldedatum haben, das vor dem Anmeldedatum der Anmeldung liegt, für die Priorität beansprucht wird.

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Prior foreign appplications
Priorität beansprucht

Priority Claimed

(Number) (Nummer)	(Country) (Land)	(Day Month Year Filed) (Tag Monat Jahr eingereicht)	Yes Ja	No Nein
19849549.8	DE	27.10.1998	☒	☐
			☐	☐
			☐	☐

Ich beanspruche hiermit gemäss Absatz 35 der Zivilprozessordnung der Vereinigten Staaten, Paragraph 120, den Vorzug aller unten aufgeführten Anmeldungen und falls der Gegenstand aus jedem Anspruch dieser Anmeldung nicht in einer früheren amerikanischen Patentanmeldung laut dem ersten Paragraphen des Absatzes 35 der Zivilprozeßordnung der Vereinigten Staaten, Paragraph 122 offenbart ist, erkenne ich gemäss Absatz 37, Bundesgesetzbuch, Paragraph 1.56(a) meine Pflicht zur Offenbarung von Informationen an, die zwischen dem Anmeldedatum der früheren Anmeldung und dem nationalen oder PCT internationalen Anmeldedatum dieser Anmeldung bekannt geworden sind.

I hereby claim the benefit under Title 35. United States Code. §120 of any United States application(s) listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States application in the manner provided by the first paragraph of Title 35, United States Code, §122, I acknowledge the duty to disclose material information as defined in Title 37, Code of Federal Regulations, §1.56(a) which occured between the filing date of the prior application and the national or PCT international filing date of this application.

(Application Serial No.) (Anmeldeseriennummer)	(Filing Date D, M, Y) (Anmeldedatum T, M, J)	(Status) (patentiert, anhangig, aufgegeben)	(Status) (patented, pending, abandoned)
PCT/DE99/03304	14.10.1999		

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I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true, and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

German Language Declaration

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